RFC3261 UAC/UAS Test Cases

For

Sip Forum Test Frame Work (SFTF)

08/18/2006

Rel. 0.1
<table>
<thead>
<tr>
<th>Rev. ID</th>
<th>Scope</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.1</td>
<td>Initial Document</td>
</tr>
</tbody>
</table>
# TABLE OF CONTENTS

1.0 LICENSE DESCRIPTION ........................................................................................................... 1

2.0 DISCLAIMER .......................................................................................................................... 1

3.0 TEST CASE DESCRIPTION .................................................................................................. 2

4.0 RFC 3261 TEST CASE EXCEPTIONS ..................................................................................... 105
1.0 LICENSE DESCRIPTION

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2.0 DISCLAIMER

This document, being an initial draft, may contain errors/bugs, which we hope contributing members to SFTF will assist in identifying and resolving.
### 3.0 TEST CASE DESCRIPTION

<table>
<thead>
<tr>
<th>#</th>
<th>RFC 3261 excerpt</th>
<th>RFC location</th>
<th>Test Case</th>
<th>Can this be combined with another clause?</th>
<th>Under Test</th>
<th>Test Case Description</th>
<th>Expected Behaviour</th>
<th>Call Flow</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>messages consist of a start-line, one or more header fields, an empty line indicating the end of the header fields.</td>
<td>section 7 page 27</td>
<td>101</td>
<td>For s.no. 1,2,3 We can write test case in two way in first test we check the presence of the start line, empty line in INVITE sent by UAC and in Second way test the behavior of UAS on missing one of these line.</td>
<td>UAS</td>
<td>In the first case as UAC we send an INVITE message to UAS with missing empty line. UAS will reject the INVITE with 400 response.</td>
<td>UAS will response back with 400 response.</td>
<td>F1: UAC-&gt;UAS INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a64b4c76e6710 CSeq: 314159 INVITE Max-Forwards: 70 Date: Thu, 21 Feb 2002 13:02:03 GMT Content-Length: 568 v=0 o=USER 2890844526 2890844526 IN IP4 here.com s=Session SDP c=IN IP4 pc33.atlanta.com t=0 0 m=audio 49172 RTP/AVP 0 a=rtpmap:0 PCMU/8000 F2: UAS-&gt;UAC SIP/2.0 400 Bad request Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a64b4c76e6710 CSeq: 314159 INVITE Content-Type: application/sdp Content-Length: 131 F3: UAC-&gt;UAS ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 Max-Forwards: 70 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td></td>
<td>start-line, each message-header line, and the empty line MUST be terminated by a carriage-return line-feed sequence (CRLF)</td>
<td>section 7 page 27</td>
<td>This section is covered in test case 101</td>
<td>UAC</td>
<td>In the second case we as UAS check the INVITE send by UAC is as per rfc3261 or not.</td>
<td>UAC will have to send INVITE as per rfc3261 format.</td>
<td>F1: UAC-&gt;UAS</td>
<td></td>
</tr>
<tr>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>call-ID: a84b4c76e66710 CSeq: 314159 ACK Content-Length: 0</td>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
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<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>Note that the empty line MUST be present even if the message-body is not.</td>
<td>section 7 page 27</td>
<td>This section is covered in test case 101</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

4 A Request-Line contains a method name, a Request-URI, and the protocol version separated by a single space. For s.no. 4, 5 and 11 we can write one single test case in two ways in first case we as

| UAC | As per RFC3261 the Request line consist of Method Request URI and SIP-Version and UAC MUST sends Request and Request-URI as per RFC 3261 format. |

F1: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
space (SP) character.  
The Request-Line ends with CRLF.  No CR or LF are allowed except in the end-of-line CRLF sequence.  No linear whitespace (LWS) is allowed in any of the elements.  
Request-Line = Method SP Request-URI SP SIP-Version CRLF

<table>
<thead>
<tr>
<th>5</th>
<th>The Request-URI</th>
<th>UAS check the whether the Request line and Request-URI in Invite of UAC is as per rfc3261 or not.</th>
<th>MUST not contain unescaped space in request URI. In first case we as UAS check the presence of Request Line and Request URI in request sends by UAC.</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>section 7</td>
<td>This section is covered in test case 102</td>
<td></td>
</tr>
</tbody>
</table>

To: Bob <sip:bob@biloxi.com> 
From: Alice <sip:alice@atlanta.com>;tag=1928301774 
Call-ID: a84b4c76e66710 
CSeq: 314159 INVITE 
Max-Forwards: 70 
Date: Thu, 21 Feb 2002 13:02:03 GMT 
Contact: <sip:alice@pc33.atlanta.com> 
Content-Length: 568

F2: UAS->UAC
SIP/2.0 603 Decline SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 Cseq: 314159 INVITE

F3: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 
Max-Forwards: 70 
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf 
From: Alice <sip:alice@atlanta.com>;tag=1928301774 
Call-ID: a84b4c76e66710 
Cseq: 314159 ACK 
Content-Length: 0

F1: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0 
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 
To: Bob <sip:bob@biloxi.com> 
From: Alice <sip:alice@atlanta.com>;tag=1928301774 
Call-ID: a84b4c76e66710 
Cseq: 314159 INVITE 
Max-Forwards: 70 
Date: Thu, 21 Feb 2002 13:02:03 GMT 
Contact: <sip:alice@pc33.atlanta.com>
<p>| | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>6</td>
<td>To be compliant with this specification, applications sending SIP messages MUST include a SIP Version of &quot;SIP/2.0&quot;. The SIP Version string is case-insensitive, but implementations MUST send upper-case.</td>
<td>section 7 page 28</td>
<td>103</td>
</tr>
<tr>
<td></td>
<td>There are three test cases for this section. One for the UAC for SIP-Version and second for UAS for SIP-Version and third for UAS with lower case Sip-Version.</td>
<td>UAC</td>
<td>As per rfc3261 all the request and response MUST contain SIP-Version as &quot;SIP/2.0&quot;. In this case we as UAS check the presence of SIP-Version in INVITE send by UAC.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>1</td>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</td>
<td>v=0</td>
<td>o=1234567890 1234567890 IN IP4 here.com</td>
</tr>
<tr>
<td></td>
<td>Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8</td>
<td>s=Session SDP</td>
<td></td>
</tr>
<tr>
<td></td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=8000</td>
<td>c=IN IP4 pc33.atlanta.com</td>
<td></td>
</tr>
<tr>
<td></td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td>t=0 0</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Call-ID: a84b4c76e66710</td>
<td>m=audio 49172 RTP/AVP 0</td>
<td></td>
</tr>
<tr>
<td></td>
<td>CSep: 314159 INVITE</td>
<td>a=rtpmap:0 PCMU/8000</td>
<td></td>
</tr>
</tbody>
</table>

F1: UAC->UAS

INVITE sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=8000
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSep: 314159 INVITE
Content-Length: 0

F2: UAS->UAC

SIP/2.0 400 Bad Request
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=8000
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSep: 314159 INVITE

F3: UAC->UAS

ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=8000
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSep: 314159 ACK
Content-Length: 0

To be compliant with this specification, applications sending SIP messages MUST include a SIP Version of "SIP/2.0". The SIP Version string is case-insensitive, but implementations MUST send upper-case.
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC
SIP/2.0 200 OK
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131

F3: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

UAS
In the second case we as UAC check the Presence of SIP-Version in BYE send by UAS

UAS MUST include SIP-Version in the BYE.

F1: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session
c=IN IP4 pc33.atlanta.com
t=0 0
<table>
<thead>
<tr>
<th>Frame</th>
<th>Source</th>
<th>Destination</th>
<th>Message Content</th>
</tr>
</thead>
<tbody>
<tr>
<td>F2: UAS-&gt;UAC</td>
<td>SIP/2.0 200 OK</td>
<td>Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8</td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE Content-Type: application/sdp Content-Length: 131</td>
</tr>
<tr>
<td>F3: UAC-&gt;UAS</td>
<td>ACK sip:bob@192.0.2.4 SIP/2.0</td>
<td>Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 Max-Forwards: 70</td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 ACK Content-Length: 0</td>
</tr>
<tr>
<td>F4: UAS-&gt;UAC</td>
<td>BYE sip:bob@192.0.2.4 SIP/2.0</td>
<td>Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 Max-Forwards: 70</td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314160 BYE Content-Length: 0</td>
</tr>
<tr>
<td>F5: UAC-&gt;UAS</td>
<td>SIP/2.0 200 OK</td>
<td>Via: SIP/2.0/UDP 192.0.2.4:branch=z9hG4bKnashds10</td>
<td>From: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf To: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 Cseq:314160 BYE Content-Length: 0</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>UAS</td>
</tr>
<tr>
<td>---</td>
<td>---</td>
<td>---</td>
<td>-----</td>
</tr>
<tr>
<td></td>
<td>In the Third case we as UAC sends INVITE message to UAS with lower SIP-Version UAS MUST accept the INVITE</td>
<td></td>
<td>F1: UAC-&gt;UAS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> sip/2.0 Via: sip/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE Max-Forwards: 70 Date: Thu, 21 Feb 2002 13:02:03 GMT Content-Length: 568</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>v=0 o=UserA 2890844526 2890844526 IN IP4 here.com s=Session SDP c=IN IP4 pc33.atlanta.com t=0 0 m=audio 49172 RTP/AVP 0 a=rtpmap:0 PCMU/8000</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>F2: UAS-&gt;UAC</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>SIP/2.0 200 OK Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE Content-Type: application/sdp Content-Length: 131</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>F3: UAC-&gt;UAS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 Max-Forwards: 70 To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 ACK Content-Length: 0</td>
</tr>
</tbody>
</table>

7 A Status-Line consists section 7 104 For s.no 7,8 and 9 UAS As per rfc3261 the 7 104 Status Line F1: UAC->UAS
of the protocol version followed by a numeric Status-Code and its associated textual phrase, with each element separated by a single SP character.

page 28

we can write one single in two way one as UAC check the format of response send by UAS and second one check the behavior of UAC mismatch Response format.

response is different from request by a status line in the first case we as UAC check the presence of status line in response from UAS.

MUST be present in Response.

<table>
<thead>
<tr>
<th>page 28</th>
<th>section 7</th>
<th>page 28</th>
<th>test case 104</th>
</tr>
</thead>
<tbody>
<tr>
<td>8</td>
<td>No CR or LF is allowed except in the final CRLF sequence.</td>
<td>This section is covered in test case 104</td>
<td>UAC</td>
</tr>
<tr>
<td></td>
<td>UAC In the Second case we as UAS send response with out</td>
<td>UAC will not understand response and</td>
<td>F1: UAC-&gt;UAS</td>
</tr>
<tr>
<td></td>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 Via: SIP/2.0/UDP</td>
<td></td>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</td>
</tr>
<tr>
<td></td>
<td>pc33.atlanta.com;branch=Z9hG4bKnashds8</td>
<td>pc33.atlanta.com;branch=Z9hG4bKnashds8</td>
<td>pc33.atlanta.com;branch=Z9hG4bKnashds8</td>
</tr>
<tr>
<td></td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
</tr>
<tr>
<td></td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td></td>
<td>Call-ID: a84b4c76e66710</td>
<td>Call-ID: a84b4c76e66710</td>
<td>Call-ID: a84b4c76e66710</td>
</tr>
<tr>
<td></td>
<td>CSeq: 314159 INVITE</td>
<td>CSeq: 314159 INVITE</td>
<td>CSeq: 314159 INVITE</td>
</tr>
<tr>
<td></td>
<td>Max-Forwards: 70</td>
<td>Max-Forwards: 70</td>
<td>Max-Forwards: 70</td>
</tr>
<tr>
<td></td>
<td>Date: Thu, 21 Feb 2002 13:02:03 GMT</td>
<td>Date: Thu, 21 Feb 2002 13:02:03 GMT</td>
<td>Date: Thu, 21 Feb 2002 13:02:03 GMT</td>
</tr>
<tr>
<td></td>
<td>Contact: <a href="">sip:alice@pc33.atlanta.com</a></td>
<td>Contact: <a href="">sip:alice@pc33.atlanta.com</a></td>
<td>Contact: <a href="">sip:alice@pc33.atlanta.com</a></td>
</tr>
<tr>
<td></td>
<td>Content-Length: 568</td>
<td>Content-Length: 568</td>
<td>Content-Length: 568</td>
</tr>
<tr>
<td></td>
<td>v=0</td>
<td>v=0</td>
<td>v=0</td>
</tr>
<tr>
<td></td>
<td>o=UserA 2890844526 2890844526 IN IP4 here.com</td>
<td>o=UserA 2890844526 2890844526 IN IP4 here.com</td>
<td>o=UserA 2890844526 2890844526 IN IP4 here.com</td>
</tr>
<tr>
<td></td>
<td>s=Session SDP</td>
<td>s=Session SDP</td>
<td>s=Session SDP</td>
</tr>
<tr>
<td></td>
<td>c=IN IP4 pc33.atlanta.com</td>
<td>c=IN IP4 pc33.atlanta.com</td>
<td>c=IN IP4 pc33.atlanta.com</td>
</tr>
<tr>
<td></td>
<td>t=0 0</td>
<td>t=0 0</td>
<td>t=0 0</td>
</tr>
<tr>
<td></td>
<td>m=audio 49172 RTP/AVP 0</td>
<td>m=audio 49172 RTP/AVP 0</td>
<td>m=audio 49172 RTP/AVP 0</td>
</tr>
<tr>
<td></td>
<td>a=rtpmap:0 PCMU/8000</td>
<td>a=rtpmap:0 PCMU/8000</td>
<td>a=rtpmap:0 PCMU/8000</td>
</tr>
<tr>
<td></td>
<td>F2: UAS-&gt;UAC</td>
<td>F2: UAS-&gt;UAC</td>
<td>F2: UAS-&gt;UAC</td>
</tr>
<tr>
<td></td>
<td>SIP/2.0 200 OK Via: SIP/2.0/UDP</td>
<td>SIP/2.0 200 OK Via: SIP/2.0/UDP</td>
<td>SIP/2.0 200 OK Via: SIP/2.0/UDP</td>
</tr>
<tr>
<td></td>
<td>pc33.atlanta.com;branch=Z9hG4bKnashds8</td>
<td>pc33.atlanta.com;branch=Z9hG4bKnashds8</td>
<td>pc33.atlanta.com;branch=Z9hG4bKnashds8</td>
</tr>
<tr>
<td></td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
</tr>
<tr>
<td></td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td></td>
<td>Call-ID: a84b4c76e66710</td>
<td>Call-ID: a84b4c76e66710</td>
<td>Call-ID: a84b4c76e66710</td>
</tr>
<tr>
<td></td>
<td>CSeq: 314159 INVITE</td>
<td>CSeq: 314159 INVITE</td>
<td>CSeq: 314159 INVITE</td>
</tr>
<tr>
<td></td>
<td>Content-Type: application/sdp Content-Length: 131</td>
<td>Content-Type: application/sdp Content-Length: 131</td>
<td>Content-Type: application/sdp Content-Length: 131</td>
</tr>
<tr>
<td></td>
<td>F3: UAC-&gt;UAS</td>
<td>F3: UAC-&gt;UAS</td>
<td>F3: UAC-&gt;UAS</td>
</tr>
<tr>
<td></td>
<td>ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP</td>
<td>ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP</td>
<td>ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP</td>
</tr>
<tr>
<td></td>
<td>pc33.atlanta.com;branch=Z9hG4bKnashds8 Max-Forwards: 70</td>
<td>pc33.atlanta.com;branch=Z9hG4bKnashds8 Max-Forwards: 70</td>
<td>pc33.atlanta.com;branch=Z9hG4bKnashds8 Max-Forwards: 70</td>
</tr>
<tr>
<td></td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
</tr>
<tr>
<td></td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td></td>
<td>Call-ID: a84b4c76e66710</td>
<td>Call-ID: a84b4c76e66710</td>
<td>Call-ID: a84b4c76e66710</td>
</tr>
<tr>
<td></td>
<td>CSeq: 314159 ACK Content-Length: 0</td>
<td>CSeq: 314159 ACK Content-Length: 0</td>
<td>CSeq: 314159 ACK Content-Length: 0</td>
</tr>
</tbody>
</table>
status line. UAC will not understand response and discard the message.

```
F1: UAC->UAS
SIP/2.0 487 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=1928301774
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Type: application/sdp
Content-Length: 568

v=0
o= UserA 2890844526 2890844526 IN IP4 here.com
s= Session SDP
c= IN IP4 pc33.atlanta.com
t= 0
m= audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

```
F2: UAS->UAC
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Type: application/sdp
Content-Length: 0
```

```
F3: UAC->UAS
SIP/2.0 487 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Type: application/sdp
Content-Length: 0
```

```
F4: UAS->UAC
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
```
<table>
<thead>
<tr>
<th>Section</th>
<th>Content</th>
</tr>
</thead>
</table>
| 9       | **Status-Line** = SIP-Version SP Status-Code SP Reason-Phrase CRLF  
This section is covered in test case 101 |
| 10      | This specification conforms to RFC 2234 [10] and uses only explicit whitespace and folding as an integral part of the grammar. |
| 105     | For s.no 10, 11, 12, 13, 14, 16 and 24 can be combine in a single test case. Here we as UAC check whether the UAS support these format or not by sending Invite with this specific format.  
For this multi-purpose test we as UAC sends an INVITE to UAS with SIP URI format, arbitrary amount of whitespace on either side of the colon, multiple lines by preceding each extra line with at least one SP or horizontal tab(HT) and multiple header field rows into one "field-name: field-value" pair, without changing the semantics of the message, by appending each subsequent field-value to the first, each separated by a comma. UAS MUST accept the INVITE and response back with 180 ringing/200 ok.  
F1: UAC->UAS  
INVITE sip:bob@biloxi.com SIP/2.0  
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8  
To: Bob <sip:bob@biloxi.com;firstparameter;secondparameter;thirdparameter>  
From: Alice <sip:alice@atlanta.com; this is a long append param with break line>;tag=a6c85cf  
Call-ID: a84b4c76e66710  
CSeq: 314159 INVITE  
Max-Forwards: 70  
Date: Thu, 21 Feb 2002 13:02:03 GMT  
Contact: <sip:alice@pc33.atlanta.com>  
Content-Type: application/sdp  
Content-Length: 568  
v=0  
o=甲方 289084526 289084526 IN IP4 here.com  
s=Session SDP  
c=IN IP4 pc33.atlanta.com  
t=0 0  
m=audio 49172 RTP/AVP 0  
a=rtpmap:0 PCMU/8000  
F2: UAS->UAC  
SIP/2.0 200 OK  
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8  
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf  
From: Alice <sip:alice@atlanta.com>;tag=1928301774  
Call-ID: a84b4c76e66710  
CSeq: 314159 INVITE  
Contact: <sip:alice@pc33.atlanta.com>  
Content-Type: application/sdp  
Content-Length: 131  
F3: UAC->UAC  
ACK sip:bob@biloxi.com SIP/2.0  
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 |
<p>| 11 | Section 25 allows for an arbitrary amount of whitespace on either side of the colon; | section 7.3 page 29 | This section is covered in test case 105 |
| 12 | Header fields can be extended over multiple lines by preceding each extra line with at least one SP or horizontal tab (HT). The line break and the whitespace at the beginning of the next line are treated as a single SP character | section 7.3.1 page 30 | This section is covered in test case 105 |
| 13 | Header fields can be extended over multiple lines by preceding each extra line with at least one SP or horizontal tab (HT). The line break and the whitespace at the beginning of the next line are treated as a single SP character | section 7.3.1 page 30 | This section is covered in test case 105 |
| 14 | It MUST be possible to combine the multiple header field rows into one &quot;field-name: field-value&quot; pair, without changing the semantics of the message, by appending each subsequent field-value to the first, each separated by a comma. | section 7.3.1 page 30 | This section is covered in test case 105 |
| 15 | The line break and the whitespace at the beginning of the next | section 7.3.1 page 30 | This section is covered in test case 105 |</p>
<table>
<thead>
<tr>
<th></th>
<th>Implementations MUST be able to process multiple header field rows with the same name in any combination of the single-value-per-line or comma-separated value forms.</th>
</tr>
</thead>
<tbody>
<tr>
<td>16</td>
<td>section 7.3.1 page 31 This section is covered in test case 105</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th></th>
<th>Even though an arbitrary number of parameter pairs may be attached to a header field value, any given parameter-name MUST NOT appear more than once.</th>
</tr>
</thead>
<tbody>
<tr>
<td>17</td>
<td>section 7.3.1 page 32 106 Here we as UAC check the behavior of UAS on repeated parameter name of Header Field.</td>
</tr>
</tbody>
</table>

**F1: UAC->UAS**

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
e=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

**F2: UAS->UAC**

```
SIP/2.0 400 Bad request
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 131
```
When comparing header fields, field names are always case-insensitive.

Here we send an INVITE message with header field name random case(lower and upper mix). UAS MUST accept this format.

Here we sends an INVITE message as per this formate.

For s.no 18 and 19 can be combine in single test case. Here we check the behavior of UAS by sending Invite message as per this formate.

UAS MUST accept the INVITE and response back with 180 ringing/200 ok.
| 19 | Unless otherwise stated in the definition of a particular header field, field values, parameter names, and parameter values are case-insensitive. Tokens are always case-insensitive. Unless specified otherwise, values expressed as quoted strings are case-sensitive | section 7.3.1 page 32 | This section is covered in test case 107 |

<p>| 20 | If a header field appears in a message not matching its category (such as a request header field in a response), it MUST be ignored. | section 7.3.2 page 32 | Here we check the behavior of UAC on getting a request with response. As per rfc3261 if a header field appears in a message not matching its category (such as a request header field in a response), it MUST be ignored. Here we as UAS sends a request with response. UAC MUST ignore it. | UAC MUST ignore the Request Header Field. |</p>
<table>
<thead>
<tr>
<th>Section</th>
<th>Line</th>
<th>Content</th>
</tr>
</thead>
<tbody>
<tr>
<td>7.3.3</td>
<td>109</td>
<td>Here we check whether the UAS support I the header field name in both short and long form or not.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>UAS</td>
</tr>
<tr>
<td></td>
<td></td>
<td>As per rfc3261 field name MAY appear in both long and short forms within the same message. Implementations MUST accept both the long and short forms of each header name. Here we as UAC sends INVITE with all Header Field name in short form. UAS MUST accept the INVITE.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>UAS MUST accept the INVITE and response back with 180 ringing/200 ok.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>F1: UAC-&gt;UAS</td>
</tr>
<tr>
<td></td>
<td></td>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 t: Bob<a href="">sip:bob@biloxi.com</a> To: Bob<a href="">sip:bob@biloxi.com</a> f: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE Max-Forwards: 70 Contact: <a href="">sip:alice@pc33.atlanta.com</a> m: <a href="">sip:alice@pc33.atlanta.com</a> l: 568 v=0 o= UserA 2890844526 2890844526 IN IP4 here.com s=Session SDP c=IN IP4 pc33.atlanta.com t=0 0 m=audio 49172 RTP/AVP 0 a=rtpmap:0 PCMU/8000</td>
</tr>
</tbody>
</table>
The Internet media type of the message body MUST be given by the Content-Type header field. If the body has undergone any encoding such as compression, then this MUST be indicated by the Content-Encoding header field; otherwise, Content-Encoding MUST be omitted.

<table>
<thead>
<tr>
<th>Section</th>
<th>Page</th>
<th>Line</th>
<th>Line Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>7.4.1</td>
<td>33</td>
<td>110</td>
<td>For s.no 22 and 115 we as UAC check the behavior on missing one Content-Encoding field while sending a encoded message.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>140</td>
<td>UAS As per rfc3261 if the body has undergone any encoding such as compression, then this MUST be indicated by the Content-Encoding header field. In this case, we as UAC sends an INVITE message with coded message Body and without Content-Encoding Header Field. UAS will not understand coding and response back with 415.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>150</td>
<td>UAS will not understand coding and response back with 415.</td>
</tr>
</tbody>
</table>

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 568

UAS

v=0
o=*
 s=*****
c=urn:ietf:params:xml:ns:base
m=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131

F3: UAC->UAS

ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F2: UAS->UAC

SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131

F3: UAC->UAS

ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=*
 s=*****
c=urn:ietf:params:xml:ns:base
m=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131

F3: UAC->UAS

ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=*
 s=*****
c=urn:ietf:params:xml:ns:base
m=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131

F3: UAC->UAS

ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
In the second case we check the behaviour of UAS on receiving INVITE without coded body but with Content-Encoding Field.

UAS will reject the INVITE with 415.

---

SIP/2.0 415
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131

F3: UAC->UAS

ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

---

In the second case we check the behaviour of UAS on receiving INVITE without coded body but with Content-Encoding Field.

UAS will reject the INVITE with 415.

---

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568
Content-Encoding: gzip

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 415
Via: SIP/2.0/UDP
| 23 | The Content-Length header field value is used to locate the end of each SIP message in a stream. It will always be present when SIP messages are sent over stream-oriented transports. | section 7.5 page 34 | 111 | Here in the First case we check the presence of Content-length in invite when send over TCP network. |

**UAC**

As per rfc3261 The Content-Length header field value is used to locate the end of each SIP message in a stream. It will always be present when SIP messages are sent over stream-oriented transports. Here we as UAS check the presence of Content-Length in INVITE when sends on TCP network. **UAC MUST include Content-Length header Field,**

**F1: UAC->UAS**

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@atlanta.com>;tag=1928301774
Max-Forwards: 70
CSeq: 314159 ACK
Content-Length: 568

v=0 o=USERA 2890844526 2890844526 IN IP4 here.com s=Session SDP c=IN IP4 pc33.atlanta.com t=0 0 m=audio 49172 RTP/AVP 0 a=rtpmap:0 PCMU/8000

**F2: UAS->UAC**

SIP/2.0 200 ok
Via: SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 131

F3: UAC->UAS

ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
In the second case we as UAC send INVITE with body but with content-length value zero:

<table>
<thead>
<tr>
<th>F1: UAC-&gt;UAS</th>
</tr>
</thead>
<tbody>
<tr>
<td>UAS reject the INVITE with 415</td>
</tr>
<tr>
<td>F2: UAS-&gt;UAC</td>
</tr>
</tbody>
</table>

| From: Alice <sip:alice@atlanta.com>;tag=1928301774 |
| Call-ID: a84b4c76e66710 |
| CSeq: 314159 INVITE |
| Contact: <sip:bob@192.0.2.4> |
| Content-Type: application/sdp |
| Content-Length: 131 |

| From: Alice <sip:alice@atlanta.com>;tag=1928301774 |
| Call-ID: a84b4c76e66710 |
| CSeq: 314159 ACK |
| Content-Length: 0 |

| From: Bob <sip:bob@biloxi.com> |
| Call-ID: a6c85cf |
| CSeq: 314159 INVITE |
| Content-Length: 0 |

<table>
<thead>
<tr>
<th>F3: UAC-&gt;UAS</th>
</tr>
</thead>
<tbody>
<tr>
<td>ACK sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</td>
</tr>
<tr>
<td>Via: SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bKnashds8</td>
</tr>
<tr>
<td>Max-Forwards: 70</td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
</tr>
<tr>
<td>CSeq: 314159 ACK</td>
</tr>
<tr>
<td>Content-Length: 0</td>
</tr>
</tbody>
</table>

| From: Alice <sip:alice@biloxi.com>;tag=1928301774 |
| Call-ID: a84b4c76e66710 |
| CSeq: 314159 ACK |
| Content-Length: 0 |

| From: Bob <sip:bob@biloxi.com> |
| Call-ID: a6c85cf |
| CSeq: 314159 INVITE |
| Content-Length: 0 |

<table>
<thead>
<tr>
<th>F1: UAC-&gt;UAS</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</td>
</tr>
<tr>
<td>Via: SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bKnashds8</td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
</tr>
<tr>
<td>CSeq: 314159 INVITE</td>
</tr>
<tr>
<td>Max-Forwards: 70</td>
</tr>
<tr>
<td>Date: Thu, 21 Feb 2002 13:02:03 GMT</td>
</tr>
<tr>
<td>Contact: <a href="">sip:alice@pc33.atlanta.com</a></td>
</tr>
<tr>
<td>Content-Length: 0</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>F2: UAS-&gt;UAC</th>
</tr>
</thead>
<tbody>
<tr>
<td>v=0</td>
</tr>
<tr>
<td>o=UserA 2890844526 2890844526 IN IP4 here.com</td>
</tr>
<tr>
<td>s=Session SDP</td>
</tr>
<tr>
<td>c=IN IP4 pc33.atlanta.com</td>
</tr>
<tr>
<td>t=0 0</td>
</tr>
<tr>
<td>m=audio 49172 RTP/AVP 0</td>
</tr>
<tr>
<td>a=rtpmap:0 PCMU/8000</td>
</tr>
</tbody>
</table>

| From: Alice <sip:alice@atlanta.com>;tag=1928301774 |
| Call-ID: a84b4c76e66710 |
| CSeq: 314159 ACK |
| Content-Length: 0 |

<p>| | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>24</td>
<td><strong>All SIP implementations MUST support the SIP URI scheme.</strong></td>
<td><strong>This section is covered in test case 105</strong></td>
<td></td>
</tr>
<tr>
<td>25</td>
<td><strong>A request outside of a dialog MUST NOT contain a To tag; the tag in the To field of a request identifies the peer of the dialog. Since no dialog is established, no tag is present.</strong></td>
<td><strong>section 8.1.1.2 page 36</strong></td>
<td></td>
</tr>
</tbody>
</table>

### F3: UAC->UAS

ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/TCP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

### UAC

As per rfc3261 a request outside of a dialog MUST NOT contain a To tag; the tag in the To field of a request identifies the peer of the dialog. Since no dialog is established, no tag is present. In the first case we as UAS check the presence of To tag in INVITE send by UAC. UAC MUST not sends To-tag in INVITE.

### UAC MUST not sends To-tag in INVITE

**F1: UAC->UAS**

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

**F2: UAS->UAC**

SIP/2.0 200 ok
Via: SIP/2.0/UDP
<table>
<thead>
<tr>
<th>PC33.atlanta.com:branch=z9hG4bKnashds8</th>
<th>pc33.atlanta.com:branch=z9hG4bKnashds8</th>
</tr>
</thead>
<tbody>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
<td>Call-ID: a84b4c76e66710</td>
</tr>
<tr>
<td>CSeq: 314159 INVITE</td>
<td>CSeq: 314159 INVITE</td>
</tr>
<tr>
<td>Contact: <a href="">sip:bob@192.0.2.4</a></td>
<td>Contact: <a href="">sip:bob@192.0.2.4</a></td>
</tr>
<tr>
<td>Content-Type: application/sdp</td>
<td>Content-Type: application/sdp</td>
</tr>
<tr>
<td>Content-Length: 131</td>
<td>Content-Length: 131</td>
</tr>
</tbody>
</table>

UAS

In the Second case we as UAC sends an INVITE containing To-tag. UAS MUST reject the INVITE with 481.

F3: UAC->UAS

ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

UAS MUST reject the INVITE with 481.

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 481 Call transition does not exist
Via: SIP/2.0/UDP pc33.atlanta.com:branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
<p>| | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>26</td>
<td>A UAC SHOULD use the display name &quot;Anonymous&quot;, along with a syntactically correct, but otherwise meaningless URI (like sip:<a href="mailto:thsis@anonymous.invalid">thsis@anonymous.invalid</a>), if the identity of the client is to remain hidden.</td>
<td>section 8.1.1.3 page 37</td>
<td>UAS We can write Test case for UAS.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>UAS As per rfc3261 UAC SHOULD use the display name &quot;Anonymous&quot;, along with a syntactically correct, but otherwise meaningless URI (like sip:<a href="mailto:thsis@anonymous.invalid">thsis@anonymous.invalid</a>), if the identity of the client is to remain hidden. UAS response back with 180 ringing/ 200 ok.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

F3: UAC->UAS
ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159
Content-Type: application/sdp
Content-Length: 131

F1: UAC->UAS
INVITE sip:thsis@anonymous.invalid SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:thsis@anonymous.invalid>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159
Content-Type: application/sdp
Content-Length: 131
<table>
<thead>
<tr>
<th>S.NO</th>
<th>Message</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>27</td>
<td>The Call-ID header field acts as a unique identifier to group together a series of messages. It MUST be the same for all requests and responses sent by either UA in a dialog.</td>
<td>section 8.1.1.4 page 37</td>
</tr>
</tbody>
</table>

For s.no 27, 29, and 31 we check whether the Call-ID is same throughout Session. This test can be done in two ways. One as UAC to check whether UAS maintain same Call-ID throughout session, second as UAS to check whether UAC maintain same Call-ID throughout session.

UAS as per RFC3261 The Call-ID header field acts as a unique identifier to group together a series of messages. It MUST be the same for all requests and responses sent by either UA in a dialog. Here we as UAC sends an INVITE message to UAS. On 200 ok response from UAS we ask UAS to send BYE and check the whether it use same Call-ID Header field value or not. UAS MUST maintain same Call-ID Field value throughout session.

UAS MUST maintain same Call-ID Field value throughout session.

F3: UAC->UAS
ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F1: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

F2: UAS->UAC
SIP/2.0 200 ok
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131

F3: UAC->UAS

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<table>
<thead>
<tr>
<th>F1: UAC-&gt;UAS</th>
<th>UAC</th>
<th>F2: UAC-&gt;UAS</th>
<th>UAC</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</td>
<td>In the second case, we as UAS check whether the UAC maintain same Call-ID sequence throughout the session. Here UAC send an INVITE message to UAS We as UAS response back with 200 ok and ask UAC to send BYE</td>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</td>
<td>UAC MUST maintain same Call-ID Field value throughout session.</td>
</tr>
<tr>
<td>Via: SIP/2.0/UDP</td>
<td></td>
<td>Via: SIP/2.0/UDP</td>
<td></td>
</tr>
<tr>
<td>pc33.atlanta.com;branch=z9hG4bKnashds8</td>
<td></td>
<td>pc33.atlanta.com;branch=z9hG4bKnashds8</td>
<td></td>
</tr>
<tr>
<td>Max-Forwards: 70</td>
<td></td>
<td>Max-Forwards: 70</td>
<td></td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
<td></td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
<td></td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td></td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td></td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
<td></td>
<td>Call-ID: a84b4c76e66710</td>
<td></td>
</tr>
<tr>
<td>CSeq: 314159 INVITE</td>
<td></td>
<td>CSeq: 314159 INVITE</td>
<td></td>
</tr>
<tr>
<td>Content-Type: application/sdp</td>
<td></td>
<td>Content-Type: application/sdp</td>
<td></td>
</tr>
<tr>
<td>Content-Length: 568</td>
<td></td>
<td>Content-Length: 131</td>
<td></td>
</tr>
</tbody>
</table>

ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F4: UAS->UAC
BYE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160 BYE

F5: UAC->UAS
SIP/2.0 200 ok
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160 BYE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131
and check the presence of same Call-Id in it. UAC MUST maintain same call-ID in through out the session.

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
da=rtpmap:0 PCMU/8000

F2: UAS->UAC
SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131

F3: UAC->UAS
ACK sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F4: UAC->UAS
BYE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160 BYE

F5: UAS->UAC
SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8

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<p>| | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>28</td>
<td>Call-ID SHOULD be the same in each registration from a UA.</td>
<td>section 8.1.1.4 page 37</td>
<td>115</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>UAC</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>The Test is same as case116 but here method is Registration. Here we as UAS check whether the UAC use same Call-ID in all Registration request. We as UAS sends a 401(Authentication require) for First registration request of UAC. On second request we check whether it use same Call-ID field value or not. UAC MUST use same Call-ID field value in second registration request.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>UAC MUST use same Call-ID field value in second registration request.</td>
</tr>
<tr>
<td></td>
<td>F1: UAC-&gt;UAS</td>
<td>REGISTER sip:registrar.biloxi.com SIP/2.0 Via:SIP/2.0/UDPbobspc.biloxi.com:5060;branch=z9hG4bKnashds7 Max-Forwards: 70 To: Bob <a href="">sip:bob@biloxi.com</a> From: Bob <a href="">sip:bob@biloxi.com</a> Call-ID: 843817637684230@998sdasdh09 CSeq: 1826 REGISTER Contact: <a href="">sip:bob@192.0.2.4</a> Expires: 7200 Content-Length: 0</td>
<td></td>
</tr>
<tr>
<td></td>
<td>F2: UAS-&gt;UAC</td>
<td>SIP/2.0 401 Authentication requir Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7 received=192.0.2.4 To: Bob <a href="">sip:bob@biloxi.com</a> ,tag=2493k59kd From: Bob <a href="">sip:bob@biloxi.com</a> ,tag=456248 Call-ID: 843817637684230@998sdasdh09 CSeq: 1826 REGISTER Contact: <a href="">sip:bob@192.0.2.4</a> Expires: 7200 Content-Length: 0</td>
<td></td>
</tr>
<tr>
<td></td>
<td>F3: UAC-&gt;UAS</td>
<td>REGISTER sip:registrar.biloxi.com SIP/2.0 Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7 .received=192.0.2.4 To: Bob <a href="">sip:bob@biloxi.com</a> ,tag=456248 From: Bob <a href="">sip:bob@biloxi.com</a> ,tag=456248 Call-ID: 843817637684230@998sdasdh09 CSeq: 1827 REGISTER Contact: <a href="">sip:bob@192.0.2.4</a> Expires: 7200</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>---</td>
<td>---</td>
<td>---</td>
<td>---</td>
</tr>
<tr>
<td>29</td>
<td>In a new request created by a UAC outside of any dialog, the Call-ID header field MUST be selected by the UAC as a globally unique identifier over space and time unless overridden by method-specific behavior.</td>
<td>section 8.1.1.4 page 37</td>
<td>This section is covered in test case 114</td>
</tr>
<tr>
<td>30</td>
<td>when requests are retried after certain failure responses that solicit an amendment to a request (for example, a challenge for authentication), these retried requests are not considered new requests, and therefore do not need new Call-ID header fields; see Section 8.1.3.5</td>
<td>section 8.1.1.4 page 38</td>
<td>This section is covered in test case 115</td>
</tr>
<tr>
<td>31</td>
<td>Call-IDs are case-sensitive and are simply compared byte-by-byte.</td>
<td>section 8.1.1.4 page 38</td>
<td>This section is covered in test case 114</td>
</tr>
<tr>
<td>32</td>
<td>The CSeq header field serves as a way to identify and order transactions. It consists</td>
<td>section 8.1.1.5 page 38</td>
<td>116 We can write test case in two way one as UAC and check whether the UAS As per rfc3261 the CSeq header field serves as a way to identify and order UAS MUST use same Cseq in response and same method in</td>
</tr>
</tbody>
</table>

Content-Length: 0
F4: UAS->UAC
SIP/2.0 200 ok
Via: SIP/2.0/UDP
bobspc.biloxi.com:5060;branch=z9hG4bKnashds7;received=192.0.2.4
To: Bob <sip:bob@biloxi.com>;tag=2493k59kd
From: Bob <sip:bob@biloxi.com>;tag=456248
Call-ID: 843817637684230@998sdasdh09
CSeq: 1826 REGISTER
Contact: <sip:bob@192.0.2.4>
Expires: 7200
Content-Length: 0
of a sequence number and a method. The method MUST match that of the request. 

UAS maintain the same Cseq during one transition and second as UAS check whether it maintain same Cseq during one transition transactions. It consists of a sequence number and a method. The method MUST match that of the request. In the first case we as UAC check whether the UAS maintain came Cseq in response. We as UAC sends an INVITE message to UAS. On 200 ok response we check the Cseq no. UAS MUST use same Cseq in response and Mehtod.

33 A UAC MUST insert a Max-Forwards header field into each request it originates with a value that SHOULD be 70.

section 8.1.1.6 page 38

We as UAC send an invite without Max-forward header field.

As per rfc3261 a UAC MUST insert a Max-Forwards header field into each request it originates with a value that SHOULD be 70. Here we as UAC

UAS response back with 400.
| send INVITE without Max-forward Header Field. | Date: Thu, 21 Feb 2002 13:02:03 GMT  
Contact: <sip:alice@pc33.atlanta.com>  
Content-Length: 568 |  
v=0  
o=UserA 2890844526 2890844526 IN IP4 here.com  
s=Session SDP  
c=IN IP4 pc33.atlanta.com  
t=0 0  
r=audio 49172 RTP/AVP 0  
a=rtpmap:0 PCMU/8000  
F2: UAS -> UAC  
SIP/2.0 400 Bad request  
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8  
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf  
From: Alice <sip:alice@atlanta.com>;tag=1928301774  
Call-ID: a84b4c76e66710  
CSeq: 314159 ACK |  
 Content-Length: 0 |  
F3: UAC -> UAS  
ACK sip:bob@192.0.2.4 SIP/2.0  
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8  
Max-Forwards: 70  
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf  
From: Alice <sip:alice@atlanta.com>;tag=1928301774  
Call-ID: a84b4c76e66710  
CSeq: 314159 ACK |  
Content-Length: 0 |

| When the UAC creates a request, it MUST insert a Via into that request. The protocol name and protocol version in the header field MUST be SIP and 2.0, respectively. | section 8.1.1.7 page 38 |  
For s.no 34 and 35 we can write one Test case in two ways: one as UAC and check whether the response sends by UAS contains Via and second one as UAS to check whether UAC Invite contains Via or not. | UAC | UAC MUST include Via Header field in INVITE with protocol version and branch parameter. | F1: UAC -> UAS  
INVITE sip:bob@biloxi.com SIP/2.0  
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8  
To: Bob <sip:bob@biloxi.com>  
From: Alice <sip:alice@atlanta.com>;tag=1928301774  
Call-ID: a84b4c76e66710  
CSeq: 314159 INVITE  
Max-Forwards: 70  
Date: Thu, 21 Feb 2002 13:02:03 GMT  
Contact: <sip:alice@pc33.atlanta.com>  
Content-Length: 568 |  
v=0 |
<p>| | | | | |</p>
<table>
<thead>
<tr>
<th></th>
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</tr>
</thead>
<tbody>
<tr>
<td>35</td>
<td>The Via header field value MUST contain a branch parameter.</td>
<td>section 8.1.1.7 page 38</td>
<td>This section is covered in test case 118</td>
<td></td>
</tr>
<tr>
<td>36</td>
<td>The Contact header field MUST be present and contain exactly one SIP or SIPS URI in any request that can result in the establishment of a dialog. For the methods defined in this specification, that includes only the INVITE request. For these requests, the scope of the Contact is global. That is, the Contact header field value contains the URI.</td>
<td>section 8.1.1.8 page 40</td>
<td>For s.no 36, 37 and 77 we can write one single test case that will check the presence of contact length and presence of URI in it.</td>
<td></td>
</tr>
</tbody>
</table>

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<td></td>
</tr>
</tbody>
</table>

F1: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Content-Length: 568
Date: Thu, 21 Feb 2002 13:02:03 GMT
V=0
o=UAS 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC
SIP/2.0 603 Decline
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Content-Length: 0

F3: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
at which the UA would like to receive requests, and this URI MUST be valid even if used in subsequent requests outside of any dialogs.

Header field and SIP-URI in it.

| s=Session SDP |
| c=IN IP4 pc33.atlanta.com |
| t=0 0 |
| m=audio 49172 RTP/AVP 0 |
| a=rtpmap:0 PCMU/8000 |

F2: UAS->UAC

sip/2.0 603 Decline
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS

ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

37 If the Request-URI or top Route header field value contains a SIPS URI, the Contact header field MUST contain a SIPS URI as well.

section 8.1.1.8 page 40

This section is covered in test case 119

38 When a timeout error is received from the transaction layer, it MUST be treated as if a 408 (Request Timeout) status code has been received.

section 8.1.3.1 page 42

It is not possible to write test case for it as how we as UAS come to know that UAC treat a timeout error as 408.

39 If a fatal transport error is reported by the transport layer (generally, due to fatal ICMP errors in UDP or

section 8.1.3.1 page 42

It is not possible to write test case for it as how we as UAS come to know that UAC treat a fatal transport error as 503.
| 40 | A UAC MUST treat any final response it does not recognize as being equivalent to the x00 response code of that class, and MUST be able to process the x00 response code for all classes. Example, if a UAC receives an unrecognized response code of 431, it can safely assume that there was something wrong with its request and treat the response as if it had received a 400 (Bad Request) response code. | section 8.1.3.2 page 42 | 120 | We can write Test case for UAC. | UAC | As per rfc3261A, UAC MUST treat any final response it does not recognize as being equivalent to the x00 response code of that class, and MUST be able to process the x00 response code for all classes. Here, we as UAS sends an error response 431 which UAC does not recognize and treat it as 400 response code. | UAC does not recognize the response consider it as 400 and terminate the session. |

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UA 2890844526 2890844526 IN IP4 her.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 432
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS

ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
| 41 | A UAC MUST treat any provisional response different than 100 that it does not recognize as 183 (Session Progress). A UAC MUST be able to process 100 and 183 responses. | section 8.1.3.2 page 42 | 121 | We can write Test case for UAC. | UAC | As per rfc3261 a UAC MUST treat any provisional response different than 100 that it does not recognize as 183 (Session Progress). Here, we as UAS sends a 170 response which UAC does not recognise and treat it as 183 and wait for 200 ok response. | UAC does not recognise the response consider it as 183 and wait for 200 ok response |

| F1: UAC->UAS |

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Cseq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 170
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAS->UAC

SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F4: UAC->UAS

ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
<table>
<thead>
<tr>
<th>Section</th>
<th>If more than one Via header field value is present in a response, the UAC SHOULD discard the message.</th>
<th>section 8.1.3.3 page 43</th>
<th>122</th>
<th>We can write Test case for UAC.</th>
<th>UAC</th>
<th>As per rfc3261 if more than one Via header field value is present in a response, the UAC SHOULD discard the message. Here we as UAS add one more via Header field value in response. On receiving the response the UAC should consider that response is misrouted and terminate the session.</th>
<th>UAC should consider that response is miss routed and terminate the session.</th>
</tr>
</thead>
<tbody>
<tr>
<td>42</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F1: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC
SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;pc35.atlanta.com;branch=z9hG4bKnashds8 Via: SIP/2.0/UDP pc56.hsc.com;branch=z9hG4babcd1234
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS
SIP/2.0 487
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
Contact: <sip:alice@pc33.atlanta.com,alice@pc33.atlanta.com>
| 43 | Upon receipt of a redirection response (for example, a 301 response status code), clients SHOULD use the URI(s) in the Contact header field to formulate one or more new requests based on the redirected request. |
| 8.1.3.4 | section 8.1.3.4 page 43 |
| 123 | For s.no 43, 44, 45, 46 and 91 we can write one test case. |
| UAC | As per rfc3261 upon receipt of a redirection response (301 response), clients SHOULD use the URI(s) in the Contact header field to formulate one or more new requests based on the redirected request and if contacting an address in the list results in a failure, as defined in the next paragraph, the element moves to the next address in the list, until the list is exhausted. If the list is exhausted, then the request has failed. Finally, once the new request has been constructed, it is sent using a new client transaction, and therefore MUST have a new branch id in via field for new transition. |

UAC MAY use URI in Contact Header field and continue with second if first result in failure and MUST use a new branch id in via field for new transition.
branch ID in the top Via field as discussed in Section 8.1.1.7. So, here we are going to test whether UAC use the URI in Contact header field and move to next listed URI in list in case of failure and finally have a new branch id in via field in new transition.

---

F3: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F4: UAC->UAS
INVITE sip: hari@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKabcd1234
To: hari@biloxi.com
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160 INVITE
Max-Forwards: 70
Date: Thu, 21 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
i=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F5: UAS->UAC
SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4babcd1234
To: Bob <sip:har@biloxi.com>;tag=a6c85cf
From: Alice<sips:alice@atlanta.com>
Call-ID: a84b4c76e66710
CSeq: 314160 INVITE

F6: UAC->UAS
ACK sip:har@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4babcd1234
Max-Forwards: 70
To: Bob <sip:har@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
<p>| | | | | |</p>
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<tbody>
<tr>
<td>44</td>
<td>a client processing 3xx class responses MUST NOT add any given URI to the target set more than once. If the original request had a SIPS URI in the Request-URI, the client MAY choose to recurse to a non-SIPS URI, but SHOULD inform the user of the redirection to an insecure URI.</td>
<td>section 8.1.3.4 page 43</td>
<td>This section is covered in test case 124</td>
<td></td>
</tr>
<tr>
<td>45</td>
<td>If contacting an address in the list results in a failure, as defined in the next paragraph, the element moves to the next address in the list, until the list is exhausted. If the list is exhausted, then the request has failed.</td>
<td>section 8.1.3.4 page 43</td>
<td>This section is covered in test case 124</td>
<td></td>
</tr>
<tr>
<td>46</td>
<td>Finally, once the new request has been constructed, it is sent using a new client transaction, and therefore MUST have a new branch ID in the top Via field as discussed in Section 8.1.1.7.</td>
<td>section 8.1.3.4 page 44</td>
<td>This section is covered in test case 124</td>
<td></td>
</tr>
<tr>
<td>47</td>
<td>the UAS MUST inspect the method of the request. If the UAS recognizes but does not support the method of a request, it MUST generate a 405(Method Not Allowed) response. Procedures for</td>
<td>section 8.2.1 page 46</td>
<td>125 For this we can write a test case to check the behavior of UAS on a request that UAS does not support and presence of Allow header field in 405.</td>
<td>UAS As per rfc3261 if the UAS recognizes but does not support the method of a request, it MUST generate a 405(Method Not Allowed) response. UAS response back with 405(method not allowed) and also add a Allow header field with it that contain all the method which UAS F1: UAC-&gt;UAS UNKNOWN sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE</td>
</tr>
</tbody>
</table>

<p>| | | | | |</p>
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<table>
<thead>
<tr>
<th>Section</th>
<th>Code</th>
<th>Support</th>
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</thead>
<tbody>
<tr>
<td>8.2.2</td>
<td>405</td>
<td>48</td>
</tr>
<tr>
<td>8.2.6</td>
<td>405</td>
<td>126</td>
</tr>
</tbody>
</table>

Generating responses are described in Section 8.2.6. The UAS MUST also add an Allow header field to the 405 (Method Not Allowed) response. The Allow header field MUST list the set of methods supported by the UAS generating the message.

Procedures for generating responses are described in Section 8.2.6. The UAS MUST also add an Allow header field to the 405 (Method Not Allowed). Here we assume UAC sends method which UAS does not support. UAS MUST response back with 405 also add Contact Header Field which shows the methods which UAS support.

If a UAS does not understand a header field in a request (that is, the header field is not defined in this specification or in any supported extension), the server MUST ignore that header field and continue processing the message.

For s.no. 48 and 49 this we can write a test case to check the behavior of UAS on a request containing a header field that UAS does not understand.

As per rfc3261 if a UAS does not understand a header field in a request (that is, the header field is not defined in this specification or in any supported extension), the server MUST ignore that header field and continue processing the request.

If a UAS does not understand a header field in a request (that is, the header field is not defined in this specification or in any supported extension), the server MUST ignore that header field and continue processing the request.

Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UAC 289084526 289084526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC
SIP/2.0 405
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
Allow:INVITE,OPTION, BYE
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
message or a UAS SHOULD ignore any malformed header fields that are not necessary for processing requests. Here we as UAC send an INVITE message to UAS with a Header field is not defined in a specification and not necessary for the processing of INVITE. UAS MUST ignore this header field and response back with 200 ok.

| 49 | A UAS SHOULD ignore any malformed header fields that are not necessary for processing requests. | section 8.2.2 page 46 | This section is covered in test case 126 |

| 50 | It is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | section 8.2.2.1 page 47 | 127 | We can write Test case for UAS. | UAS | As per rfc3261 it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | UAS accept requests even if they do not recognize the URI scheme | F1: UAC->UAS |

| | | | | | | | | |

| 51 | It is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | section 8.2.2.1 page 47 | 127 | We can write Test case for UAS. | UAS | As per rfc3261 it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | UAS accept requests even if they do not recognize the URI scheme | F1: UAC->UAS | INVITE tel:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <tel:bob@biloxi.com> From: Alice <sip:alice@atlanta.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |

| 56 | We can write Test case for UAS. | section 8.2.2.1 page 47 | 127 | We can write Test case for UAS. | UAS | As per rfc3261 it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | UAS accept requests even if they do not recognize the URI scheme | F1: UAC->UAS | INVITE tel:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <tel:bob@biloxi.com> From: Alice <sip:alice@atlanta.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |

| 57 | We can write Test case for UAS. | section 8.2.2.1 page 47 | 127 | We can write Test case for UAS. | UAS | As per rfc3261 it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | UAS accept requests even if they do not recognize the URI scheme | F1: UAC->UAS | INVITE tel:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <tel:bob@biloxi.com> From: Alice <sip:alice@atlanta.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |

| 58 | We can write Test case for UAS. | section 8.2.2.1 page 47 | 127 | We can write Test case for UAS. | UAS | As per rfc3261 it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | UAS accept requests even if they do not recognize the URI scheme | F1: UAC->UAS | INVITE tel:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <tel:bob@biloxi.com> From: Alice <sip:alice@atlanta.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |

| 59 | We can write Test case for UAS. | section 8.2.2.1 page 47 | 127 | We can write Test case for UAS. | UAS | As per rfc3261 it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | UAS accept requests even if they do not recognize the URI scheme | F1: UAC->UAS | INVITE tel:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <tel:bob@biloxi.com> From: Alice <sip:alice@atlanta.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |

| 60 | We can write Test case for UAS. | section 8.2.2.1 page 47 | 127 | We can write Test case for UAS. | UAS | As per rfc3261 it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | UAS accept requests even if they do not recognize the URI scheme | F1: UAC->UAS | INVITE tel:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <tel:bob@biloxi.com> From: Alice <sip:alice@atlanta.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |

| 61 | We can write Test case for UAS. | section 8.2.2.1 page 47 | 127 | We can write Test case for UAS. | UAS | As per rfc3261 it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | UAS accept requests even if they do not recognize the URI scheme | F1: UAC->UAS | INVITE tel:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <tel:bob@biloxi.com> From: Alice <sip:alice@atlanta.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |

| 62 | We can write Test case for UAS. | section 8.2.2.1 page 47 | 127 | We can write Test case for UAS. | UAS | As per rfc3261 it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | UAS accept requests even if they do not recognize the URI scheme | F1: UAC->UAS | INVITE tel:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <tel:bob@biloxi.com> From: Alice <sip:alice@atlanta.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |

| 63 | We can write Test case for UAS. | section 8.2.2.1 page 47 | 127 | We can write Test case for UAS. | UAS | As per rfc3261 it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | UAS accept requests even if they do not recognize the URI scheme | F1: UAC->UAS | INVITE tel:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <tel:bob@biloxi.com> From: Alice <sip:alice@atlanta.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |

| 64 | We can write Test case for UAS. | section 8.2.2.1 page 47 | 127 | We can write Test case for UAS. | UAS | As per rfc3261 it is RECOMMENDED that a UAS accept requests even if they do not recognize the URI scheme (for example, a tel: URI) in the To header field, or if the To header field does not address a known or current user of this UAS, the UAS decides to reject the request, it SHOULD generate a | UAS accept requests even if they do not recognize the URI scheme | F1: UAC->UAS | INVITE tel:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <tel:bob@biloxi.com> From: Alice <sip:alice@atlanta.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |
response with a 403 (Forbidden) status code and pass it to the server transaction for transmission.

| If the request has no tag in the To header field, the UAS core MUST check the request against ongoing transactions. If the From tag, Call-ID, and CSeq exactly match those associated with an ongoing transaction, but the request does not match that transaction (based on the matching rules in Section 17.2.3), the UAS core SHOULD generate a 482 (Loop Detected) response and pass it to the server transaction. | section 8.2.2.2 page 47 | 128 | We can write this Test for UAS. | UAS | As per rfc3261 if the request has no tag in the To header field, the UAS core MUST check the request against ongoing transactions. If the From tag, Call-ID, and CSeq exactly match those associated with an ongoing transaction, but the request does not match that transaction (based on the matching rules in Section 17.2.3), the UAS MUST response back with 482(loop dectect) | UAS | v=0 o=UAS 2890844526 2890844526 IN IP4 here.com s=Session SDP c=IN IP4 pc33.atlanta.com t=0 0 m=audio 49172 RTP/AVP 0 a=rtpmap:0 PCMU/8000 |

F2: UAS->UAC

```
SIP/2.0 403 Forbidden
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
```

F3: UAC->UAS

```
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
```

51

We can write this Test for UAS.
core SHOULD generate a 482 (Loop Detected) response and pass it to the server transaction. Here we as UAC sends INVITE to UAS. On 180 ringing we send another request (BYE) without To tag but having same From tag, Call-ID, and CSeq exactly match those associated with an ongoing transaction. UAS MUST response back with 482 (loop detected)

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
```

F2: UAS->UAC

```
SIP/2.0 180 ringing
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
```

F3: UAC->UAS

```
BYE sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 BYE
Content-Length: 0
```

F4: UAS->UAC

```
SIP/2.0 482 loop detected
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
```

F5: UAC->UAS

```
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
```

52 The same request has arrived at the UAS more than once, following different section 8.2.2.2 page 47

129 We can write this Test for UAS on receiving same request more than UAS

UAS As per rfc3261 if same request has arrived at the UAS more than once, UAS MUST response back with 482 (loop detected)

F1: UAC->UAS

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP
```

```
paths, most likely due to forking. The UAS processes the first such request received and responds with a 482 (Loop Detected) to the rest of them.

Once.

Following different paths, most likely due to forking. The UAS processes the first such request received and responds with a 482 (Loop Detected) to the rest of them. Here we as UAC sends two INVITE messages to UAS with different paths in Via header field. UAS MUST response back with 482 (loop detected)

```
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 00 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UAC 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2: UAS->UAC

```
SIP/2.0 180 ringing
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=a6c85cf
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 00 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568
```

F3: UAC->UAS

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP ab.55.hsc.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Date: Thu, 00 Feb 2002 13:02:03 GMT
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568
```

F4: UAS->UAC

```
SIP/2.0 482 Loop detected
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>
```
<table>
<thead>
<tr>
<th>53</th>
<th>The UAS MUST add an Unsupported header field, and list in it those options it does not understand amongst those in the Require header field of the request.</th>
</tr>
</thead>
<tbody>
<tr>
<td>131</td>
<td>We can write Test case for UAS.</td>
</tr>
<tr>
<td>UAS</td>
<td>As per rfc3261 the UAS MUST add an Unsupported header field, and list in it those options it does not understand amongst those in the Require header field of the request. Here we as UAC sends INVITE to UAS with option-tag listed in a Require header field which UAS does not understand. UAS MUST reject it with 420 and MUST add an Unsupported header field, and list in it those options it does not understand amongst those in the Require header field of the request.</td>
</tr>
<tr>
<td>UAS</td>
<td>UAS MUST reject it with 420 and MUST add an Unsupported header field, and list in it those options it does not understand amongst those in the Require header field of the request.</td>
</tr>
</tbody>
</table>

Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F5: UAC->UAS

ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Require: python
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAC->UAC

SIP/2.0 420
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Unsupported: python
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS
| 54 | Require and Proxy-Require MUST NOT be used in a SIP CANCEL request, or in an ACK request sent for a non-2xx response. These header fields MUST be ignored if they are present in these requests. | section 8.2.2.3 page 48 | 132 | We can write Test case for UAS and UAC in four way. | UAC | In the first case UAC sends INVITE to UAS. We as UAS sends 180 ringing and send CANCEL. We as UAS check whether the UAC use Require and Proxy-Require. UAC MUST NOT use Require and Proxy-Require | UAC MUST NOT use Require and Proxy-Require |

**F1:** UAC->UAS

ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

**F2:** UAS->UAC

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

**F3:** UAC->UAS

CANCEL sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
### Second Case

In the Second case UAC sends INVITE to UAS. we as UAS sends 200ok. UAC sends ACK we as UAS check whether it contain Require and Proxy-Require. UAC MUST not include these Header Field.

<table>
<thead>
<tr>
<th>UAC</th>
<th>UAC MUST not include these Header Field.</th>
</tr>
</thead>
<tbody>
<tr>
<td>F1: UAC-&gt;UAS</td>
<td></td>
</tr>
<tr>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</td>
<td></td>
</tr>
<tr>
<td>Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8</td>
<td></td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
<td></td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td></td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
<td></td>
</tr>
<tr>
<td>CSeq: 314159 INVITE</td>
<td></td>
</tr>
<tr>
<td>Max-Forwards: 70</td>
<td></td>
</tr>
<tr>
<td>Contact: <a href="">sip:alice@pc33.atlanta.com</a></td>
<td></td>
</tr>
<tr>
<td>Content-Length: 568</td>
<td></td>
</tr>
<tr>
<td>v=0</td>
<td></td>
</tr>
<tr>
<td>o=PeerA 2890844526 2890844526 IN IP4 here.com</td>
<td></td>
</tr>
<tr>
<td>s=Session SDP</td>
<td></td>
</tr>
<tr>
<td>c=IN IP4 pc33.atlanta.com</td>
<td></td>
</tr>
<tr>
<td>t=0 0</td>
<td></td>
</tr>
<tr>
<td>m=audio 49172 RTP/AVP 0</td>
<td></td>
</tr>
<tr>
<td>a=rtpmap:0 PCMU/8000</td>
<td></td>
</tr>
</tbody>
</table>

| F2: UAS->UAC |
| SIP/2.0 200 ok |
| Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 |
| To: Bob <sip:bob@biloxi.com> |
| From: Alice <sip:alice@atlanta.com>;tag=1928301774 |
| Call-ID: a84b4c76e66710 |
| CSeq: 314159 INVITE |

<p>| F3: UAC-&gt;UAS |
| ACK sip:bob@192.0.2.4 SIP/2.0 |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
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</thead>
<tbody>
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</tr>
</tbody>
</table>

**UAS**

In the third case we as UAC send INVITE to UAS. On 180 ringing we send CANCEL request containing Require header filed. UAS MUST ignore this header filed present in the request.

**Content-Length: 0**

**F1: UAC->UAS**

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP
d33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 d33.atlanta.com
i=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

**F2: UAS->UAC**

SIP/2.0 180 Ringing
Via:SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>

**F3: UAC->UAS**

CANCEL sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP
d33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>
Call-ID: a84b4c76e66710
CSeq: 314159 CANCEL
Max-Forwards: 70

<table>
<thead>
<tr>
<th>F1: UAC-&gt;UAS</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</td>
</tr>
<tr>
<td>Via: SIP/2.0/UDP</td>
</tr>
<tr>
<td>pc33.atlanta.com;branch=z9hG4bKnashds8</td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
</tr>
<tr>
<td>CSeq: 314159 INVITE</td>
</tr>
<tr>
<td>v=0</td>
</tr>
<tr>
<td>o=UserA 2890844526 2890844526 IN IP4 here.com</td>
</tr>
<tr>
<td>s=Session SDP</td>
</tr>
<tr>
<td>c=IN IP4 pc33.atlanta.com</td>
</tr>
<tr>
<td>t=0 0</td>
</tr>
<tr>
<td>m=audio 49172 RTP/AVP 0</td>
</tr>
<tr>
<td>a=rtcpmap:0 PCMU/8000</td>
</tr>
<tr>
<td>F2: UAS-&gt;UAC</td>
</tr>
<tr>
<td>SIP/2.0 200 ok</td>
</tr>
<tr>
<td>Via: SIP/2.0/UDP</td>
</tr>
<tr>
<td>pc33.atlanta.com;branch=z9hG4bKnashds8</td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
</tr>
<tr>
<td>CSeq: 314159 INVITE</td>
</tr>
<tr>
<td>F3: UAC-&gt;UAS</td>
</tr>
<tr>
<td>ACK sip:bob@192.0.2.4 SIP/2.0</td>
</tr>
<tr>
<td>Via: SIP/2.0/UDP</td>
</tr>
</tbody>
</table>

UAS

In the fourth case we as UAC sends INVITE to UAS. On 200 ok response we sends ACK containing Require header field. UAS MUST ignore these header field. UAS MUST ignore these header field.
<table>
<thead>
<tr>
<th>Section</th>
<th>Text</th>
<th>Page</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>8.2.3</td>
<td>The response MUST contain an Accept header field listing the types of all bodies it understands, in the event the request contained bodies of types not supported by the UAS. If the request contained content encodings not understood by the UAS, the response MUST contain an Accept-Encoding header field listing the encodings understood by the UAS. If the request contained content with languages not understood by the UAS, the response MUST contain an Accept-Language header field indicating the languages understood by the UAS.</td>
<td>48</td>
<td></td>
</tr>
<tr>
<td>133</td>
<td>We can write Test case for UAS.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>UAS</td>
<td>As per rfc3261 The response MUST contain an Accept header field listing the types of all bodies it understands, in the event the request contained bodies of types not supported by the UAS. If the request contained content encodings not understood by the UAS, the response MUST contain an Accept-Encoding header field listing the encodings understood by the UAS. If the request contained content with languages not understood by the UAS, the response MUST contain an Accept-Language header field indicating the languages understood by the UAS. Here we as UAC sends INVITE with a body whose type (indicated by the Content-Type), language</td>
<td></td>
<td></td>
</tr>
<tr>
<td>UAS</td>
<td>The response back with 415 and response MUST contain an Accept header field listing the types of all bodies it understands, in the event the request contained bodies of types not supported by the UAS.</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**F1:** UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Encoding: unknown
Content-Language: xyz
Content-Length: 568

**F2:** UAS->UAC

SIP/2.0 415
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Accept-Encoding: gzip
Accept-Language: da, en-gb;q=0.8, en;q=0.7
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

**F3:** UAC->UAS

ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
| 56 | A UAS that wishes to apply some extension when generating the response MUST NOT do so unless support for that extension is indicated in the Supported header field in the request. | section 8.2.4 page 49 | 134 | We can write Test case for UAS. | UAS As per rfc3261 UASs SHOULD generate a final response to a non-INVITE request. we as UAC sends an INVITE to UAS. on receiving 200ok, we send BYE request. UAS SHOULD generate final response(200ok) | UAS SHOULD generate final response(200ok) | F1: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC
SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Content-Length: 568
Contact: <sip:alice@pc33.atlanta.com>
SIP/2.0 200 ok
Content-Length: 568

F3: UAC->UAS
BYE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Content-Length: 568

F4: UAS->UAC
SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Content-Length: 568

| (indicated by the Content-Language) or encoding (indicated by the Content-Encoding) are not understood by UAS. UAS response back with 415 and response MUST contain an Accept header field listing the types of all bodies it understands, in the event the request contained bodies of types not supported by the UAS. | To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 CANCEL
Content-Length: 0 |
One largely non-method-specific guideline for the generation of responses is that UASs SHOULD NOT issue a provisional response for a non-INVITE request. Rather, UASs SHOULD generate a final response to a non-INVITE request as soon as possible.

We can write Test case for UAS.

As per rfc3261 UASs SHOULD generate a final response to a non-INVITE request, we as UAC sends an INVITE to UAS. on receiving 200ok, we send BYE request. UAS SHOULD generate final response(200ok)
When a 100 (Trying) response is generated, any Timestamp header field present in the request MUST be copied into this 100 (Trying) response. If there is a delay in generating the response, the UAS SHOULD add a delay value into the Timestamp value in the response. This value MUST contain the difference between the time of sending of the request and receipt of the request, measured in seconds.

<table>
<thead>
<tr>
<th>Page</th>
<th>Section</th>
<th>We can write Test case for UAS.</th>
<th>UAS SHOULD add a delay value into the Timestamp value in the response.</th>
</tr>
</thead>
<tbody>
<tr>
<td>58</td>
<td>8.2.6.2</td>
<td>136</td>
<td></td>
</tr>
</tbody>
</table>

As per rfc3261 When a 100 (Trying) response is generated, any Timestamp header field present in the request MUST be copied into this 100 (Trying) response. If there is a delay in generating the response, the UAS SHOULD add a delay value into the Timestamp value in the response. This value MUST contain the difference between the time of sending of the request and receipt of the request, measured in seconds. Here we as UAC sends INVITE to UAS with timestamp header field. UAS SHOULD add a delay value into the Timestamp value in the response.
<table>
<thead>
<tr>
<th>Page</th>
<th>Section</th>
<th>Description</th>
<th>Example</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>59</td>
<td>section 8.2.6.2 page 50</td>
<td>The From field of the response MUST equal the From header field of the request. The Call-ID header field of the response MUST equal the Call-ID header field of the request. The CSeq header field of the response MUST equal the CSeq field of the request.</td>
<td>section 8.2.6.2 page 50</td>
<td>For s.no. 59 and 79 We can write a single Test which can check whether the three fields have same value or not.</td>
</tr>
<tr>
<td>137</td>
<td>UAS</td>
<td>As per rfc3261 the from field of the response MUST equal the From header field of the request. The Call-ID header field of the response MUST equal the Call-ID header field of the request. The CSeq header field of the response MUST equal the CSeq field of the request.</td>
<td>CSeq: 314159 ACK Content-Length: 0</td>
<td>UAS MUST use same Field value in response</td>
</tr>
<tr>
<td>60</td>
<td>section 8.2.6.2 page 50</td>
<td>If a request contained a To tag in the request, the To header field in</td>
<td></td>
<td>We can write Test case for UAS.</td>
</tr>
<tr>
<td>138</td>
<td>UAS</td>
<td>As per rfc3261 if a request contained a To tag in the</td>
<td></td>
<td>UAS MUST contain same To-Tag.</td>
</tr>
</tbody>
</table>

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568
v=0
o=UAS 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS

ACK sip:bob@192.0.2.4 SIP/2.0
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
When UAC sends INVITE, it includes an INVITE request, the To header field in the response MUST equal that of the request. Here we as UAC sends INVITE to UAS. After sending ACK we sends BYE and check whether the response from UAS contains same To tag or not. UAS MUST contain same To-Tag.

<table>
<thead>
<tr>
<th>request</th>
<th>response</th>
</tr>
</thead>
</table>
| Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:bob@biloxi.com> 
From: Alice <sip:alice@atlanta.com>;tag=1928301774 
Call-ID: a84b4c76e66710 
CSeq: 314159 INVITE Max-Forwards: 70 
Contact: <sip:alice@pc33.atlanta.com> 
Content-Length: 568 |
| v=0 o=GuestC 2890844526 2890844526 IN IP4 here.com s=Session SDP c=IN IP4 pc33.atlanta.com t=0 0 m=audio 49172 RTP/AVP 0 a=rtpmap:0 PCMU/8000 |

F2: UAS->UAC
SIP/2.0 200 ok 
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf 
From: Alice <sip:alice@atlanta.com>;tag=1928301774 
Call-ID: a84b4c76e66710 
CSeq: 314159 INVITE |

F3: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0 
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 
Max-Forwards: 70 
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf 
From: Alice <sip:alice@atlanta.com>;tag=1928301774 
Call-ID: a84b4c76e66710 
CSeq: 314159 ACK 
Content-Length: 0 |

F4: UAC->UAS
BYE sip:bob@192.0.2.4 SIP/2.0 
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 
Max-Forwards: 70 
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf 
From: Alice <sip:alice@atlanta.com>;tag=1928301774 
Call-ID: a84b4c76e66710 |
If the To header field in the request did not contain a tag, the URI in the To header field in the response MUST equal the URI in the To header field; additionally, the UAS MUST add a tag to the To header field in the response.

We can write a test case for UAS to check whether it accept a request without To tag and add To tag in response.

As per rfc3261 if the To header field in the request did not contain a tag, the URI in the To header field in the response MUST equal the URI in the To header field; additionally, the UAS MUST add a tag to the To header field in the response. Here we UAC send an INVITE to UAS. After ACK we sends BYE request to UAS without To tag. UAS on response MUST add To tag in response.
<p>| | | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>62</td>
<td>In that usage, a UAS that receives a CANCEL request for an INVITE, but has not yet sent a final response, would &quot;stop ringing&quot;, and then respond to the INVITE with a specific error response (a 487).</td>
<td>section 9 page 53</td>
<td>140</td>
<td>For s.no 62 and 69 we can write Test case for UAS to test its behavior on receiving Cancel request.</td>
</tr>
</tbody>
</table>
UAS would “stop ringing”, and then respond to the INVITE with a specific error response (a 487).

F2: UAS->UAC
SIP/2.0 180 ringing
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS
CANCEL sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 CANCEL
Content-Length: 0

F4: UAS->UAC
SIP/2.0 487
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 BYE
Content-Length: 0

F5: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via:SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK

If the transaction for the original request still exists, the behavior of the UAS on receiving a CANCEL request has no effect on the processing of the original request.

For s.no 63 and 64 we write can write one single test for UAS

As per rfc3261 if the transaction for the original request still exists, the CANCEL request has no effect on the processing of the original request.
<table>
<thead>
<tr>
<th>CANCEL request depends on whether it has already sent a final response for the original request. If it has, the CANCEL request has no effect on the processing of the original request, no effect on any session state, and no effect on the responses generated for the original request.</th>
<th>behavior of the UAS on receiving a CANCEL request depends on whether it has already sent a final response for the original request. If it has, the CANCEL request has no effect on the processing of the original request, no effect on any session state, and no effect on the responses generated for the original request.</th>
<th>the original request</th>
<th>pc33.atlanta.com;branch=z9hG4bKnashds8</th>
</tr>
</thead>
<tbody>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE Max-Forwards: 70 Contact: <a href="">sip:alice@pc33.atlanta.com</a> Content-Length: 568</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>v=0 o=Peer 2890844526 2890844526 IN IP4 here.com s=Session SDP c=IN IP4 pc33.atlanta.com t=0 0 m=audio 49172 RTP/AVP 0 a=rtpmap:0 PCMU/8000</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>F2: UAS-&gt;UAC</td>
<td>SIP/2.0 200 ok Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE</td>
<td></td>
<td></td>
</tr>
<tr>
<td>F3: UAC-&gt;UAS</td>
<td>CANCEL sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 Max-Forwards: 70 To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 CANCEL Content-Length: 0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>F4 : UAS-&gt;UAC</td>
<td>SIP/2.0 400 Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Section</td>
<td>Page</td>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>---------</td>
<td>------</td>
<td>-------------</td>
<td></td>
</tr>
<tr>
<td>9.2</td>
<td>55</td>
<td>A CANCEL request has no impact on the processing of transactions with any other method defined in this specification.</td>
<td></td>
</tr>
<tr>
<td>9.2</td>
<td>55</td>
<td>Regardless of the method of the original request, as long as the CANCEL matched an existing transaction, the UAS answers the CANCEL request itself with a 200 (OK) response.</td>
<td></td>
</tr>
</tbody>
</table>

F5: UAC->UAS

ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnaashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnaashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 180 ringing
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnaashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
<p>| | | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>66</td>
<td>The CANCEL request MUST NOT contain any Require or Proxy-Require header fields.</td>
<td>section 9.1 page 54</td>
<td>143</td>
<td>we can write a test case to check the presence of Require in cancel request send by UAC.</td>
</tr>
<tr>
<td></td>
<td>UAC</td>
<td></td>
<td></td>
<td>As per rfc3261 the CANCEL request MUST NOT contain any Require or Proxy-Require header fields. Here we as UAS, ask UAC to send INVITE. After sending 180 ok response we ask UAC to send a CANCEL request. The request MUST NOT contain any Require or Proxy-Require header fields.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>The request MUST NOT contain any Require or Proxy-Require header fields.</td>
</tr>
<tr>
<td>F3:</td>
<td>UAC-&gt;UAS</td>
<td>CANCEL sip: bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 Max-Forwards: 70 To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 CANCEL Content-Length: 0</td>
<td></td>
<td></td>
</tr>
<tr>
<td>F4:</td>
<td>UAS-&gt;UAC</td>
<td>SIP/2.0 200 ok Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Licensed to SIPfoundry by Hughes Systique Corporation under a Contributor Agreement using the GNU Lesser General Public License
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice<sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS
CANCEL sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 CANCEL
Content-Length: 0

F4: UAS->UAC
SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159

67 The destination address, port, and transport for the CANCEL MUST be identical to those used to send the original request.

section 9.2 page 55

For s.no 68 and 83 we as UAC check the Behavior of UAS on miss match cancel request

UAS

As per rfc3261 The destination address, port, and transport for the CANCEL MUST be identical to those used to send the original request. Here we as UAS, ask UAC to send INVITE. After sending 180 ok response we ask UAC to send a CANCEL request. The destination address, port, and transport for the CANCEL MUST be identical to those used to send the original request. UAC send an

F1: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com:5060>
From: Alice <sip:alice@atlanta.com>
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC
<table>
<thead>
<tr>
<th>Message</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE message to UAS. We as UAS send 180 ringing. UAC sends a CANCEL request. We as UAS check whether the UAC use same port and transport or not.</td>
<td></td>
</tr>
</tbody>
</table>

### SIP/2.0 180 ringing

```
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com:5060>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
```

F3: UAC->UAS

```
CANCEL sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: b85b6c76e66710
CSeq: 314159 CANCEL
Content-Length: 0
```

F4: UAS->UAC

```
SIP/2.0 481 call transaction does not exit
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
```

F5: UAC->UAS

```
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: b85b6c76e66710
CSeq: 314159 ACK
Content-Length: 0
```

If the UAS did not find a matching transaction for the CANCEL according to the procedure above, it SHOULD respond to the CANCEL with a 481.

---

68 If the UAS did not find a matching transaction for the CANCEL according to the procedure above, it SHOULD respond to the CANCEL with a 481.

92 For s.no 68 and 83 we as UAC check the Behavior of UAS on miss match cancel request

145 UAS As per rfc3261 if the UAS did not find a matching transaction for the CANCEL according to the procedure above, it UAS MUST response back with 481.

F1: UAC->UAS

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com:5060>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
```

---

63 Licensed to SIPfoundry by Hughes Systique Corporation under a Contributor Agreement using the GNU Lesser General Public License
<table>
<thead>
<tr>
<th>SHOULd respond to the CANCEl with a 481 (Call Leg/Transaction Does Not Exist). Here we as UAC sends an INVITE message to UAS. On receiving 200 ok response we sends a CANCEL request to UAS that does not match to existing session means we change the Call-ID, Cseq field value. UAS MUST response back with 481.</th>
</tr>
</thead>
</table>

```
Call-ID: a84b4c76e66710
CSeq: 314159
INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=alice 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCM U/8000
F2: UAS->UAC
SIP/2.0 200 OK
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com:5060>;tag=a6c85cf
From: Alice<sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS
CANCEL sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com:5060>;tag=a6c85cf
From: Alice<sip:alice@atlanta.com>;tag=1928301774
Call-ID: b85b6c76e66710
CSeq: 314159 CANCEL
Content-Length: 0
F4: UAS->UAC
ACK sip:bob@192.0.2.4 SIP/2.0
```
<p>| | | | | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>69</td>
<td>If the UAS has not issued a final response for the original request, its behavior depends on the method of the original request. If the original request was an INVITE, the UAS SHOULD immediately respond to the INVITE with a 487 (Request Terminated).</td>
<td>section 9.2 page 55</td>
<td>This section is covered in test case 140</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>70</td>
<td>The Record-Route header field has no meaning in REGISTER requests or responses, and MUST be ignored if present. In particular, the UAC MUST NOT create a new route set based on the presence or absence of a Record-Route header field in any response to a REGISTER request.</td>
<td>Section 10.2 page 55</td>
<td>146</td>
<td>For s.no .70 and 73 we write can write one single test for UAC</td>
<td>UAC</td>
<td>UAC MUST NOT include Record-Route header field and MUST contain SIP URI in Contact header field.</td>
</tr>
</tbody>
</table>

**F1: UAC->UAS**

```
REGISTER sip:registrar.biloxi.com SIP/2.0
Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=456248
From: Bob <sip:bob@biloxi.com>;tag=456248
Call-ID: 843817637684230@998sdasdh09
CSeq: 1826 REGISTER
Contact: <sip:bob@192.0.2.4> Expires: 7200
Content-Length: 0
```

**F2: UAS->UAC**

```
SIP/2.0 200 ok
Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7
:received=192.0.2.4
To: Bob <sip:bob@biloxi.com>;tag=2493k59kd
From: Bob <sip:bob@biloxi.com>;tag=456248
Call-ID: 843817637684230@998sdasdh09
CSeq: 1826 REGISTER
Contact: <sip:bob@192.0.2.4> Expires: 7200
Content-Length: 0
```
<table>
<thead>
<tr>
<th>Field in SIP URI</th>
<th>UAS</th>
<th>UAC</th>
<th>F1: UAS -&gt; UAS</th>
<th>F2: UAS -&gt; UAC</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>REGISTER sip:registrar.biloxi.com SIP/2.0 Via:SIP/2.0/UDPbospc.biloxi.com:5060;branch=z9hG4bKnashds7 Max-Forwards: 70 To: Bob <a href="">sip:bob@biloxi.com</a> From: Bob <a href="">sip:bob@biloxi.com</a>;tag=456248 Record-Route: <a href="">sip:ss2.biloxi.example.com;lr</a> Call-ID: 843817637684230@998dasdh09 CSeq: 1826 REGISTER Contact: <a href="">sip:bob@192.0.2.4</a> Expires: 7200 Content-Length: 0</td>
<td>SIP/2.0 200 ok Via: SIP/2.0/UDPbobspc.biloxi.com:5060;branch=z9hG4bKnashds7;received=192.0.2.4 To: Bob <a href="">sip:bob@biloxi.com</a>;tag=2493k59kd From: Bob <a href="">sip:bob@biloxi.com</a>;tag=456248 Call-ID: 843817637684230@998dasdh09 CSeq: 1826 REGISTER Contact: <a href="">sip:bob@192.0.2.4</a> Expires: 7200 Content-Length: 0</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Field in SIP URI</th>
<th>UAS</th>
<th>UAC</th>
<th>F1: UAS -&gt; UAS</th>
<th>F2: UAS -&gt; UAC</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>REGISTER sip:registrar.biloxi.com SIP/2.0 Via:SIP/2.0/UDPbospc.biloxi.com:5060;branch=z9hG4bKnashds7 Max-Forwards: 70 To: Bob <a href="">sip:bob@biloxi.com</a> From: Bob <a href="">sip:bob@biloxi.com</a>;tag=456248 Record-Route: <a href="">sip:ss2.biloxi.example.com;lr</a> Call-ID: 843817637684230@998dasdh09 CSeq: 1826 REGISTER Contact: <a href="">sip:bob@192.0.2.4</a> Expires: 7200 Content-Length: 0</td>
<td>SIP/2.0 200 ok Via: SIP/2.0/UDPbobspc.biloxi.com:5060;branch=z9hG4bKnashds7;received=192.0.2.4 To: Bob <a href="">sip:bob@biloxi.com</a>;tag=2493k59kd From: Bob <a href="">sip:bob@biloxi.com</a>;tag=456248 Call-ID: 843817637684230@998dasdh09 CSeq: 1826 REGISTER Contact: <a href="">sip:bob@192.0.2.4</a> Expires: 7200 Content-Length: 0</td>
</tr>
</tbody>
</table>

71 The following header fields, except Contact, MUST be included in a REGISTER request. A Contact header field MAY be included: Request-URI: The Request-URI names the domain of the service for which the registration is meant (for example, "sip:chicago.com"). The "userinfo" and "@" components of the SIP URI MUST NOT be present. To: The To header field contains the section 10.2 page 58 147 We as UAS check the presence of all these header field in Register request. UAS MUST ignore the Record- route Header field. UAC As per rfc3261 REGISTER Request MUST include Request- URI, To, Call-ID, Cseq,Contact Header fields. We as UAC check the presence of these filed in Register request sends by UAS. UAC MUST include all these header fields in Request. UAC MUST include all these header fields in Request. F1: UAS -> UAS REGISTER sip:registrar.biloxi.com SIP/2.0 Via:SIP/2.0/UDPbospc.biloxi.com:5060;branch=z9hG4bKnashds7 Max-Forwards: 70 To: Bob <sip:bob@biloxi.com> From: Bob <sip:bob@biloxi.com>;tag=456248 Record-Route: <sip:ss2.biloxi.example.com;lr> Call-ID: 843817637684230@998dasdh09 CSeq: 1826 REGISTER Contact: <sip:bob@192.0.2.4> Expires: 7200 Content-Length: 0 | F2: UAS -> UAC SIP/2.0 200 ok Via: SIP/2.0/UDPbobspc.biloxi.com:5060;branch=z9hG4bKnashds7;received=192.0.2.4 To: Bob <sip:bob@biloxi.com>;tag=2493k59kd From: Bob <sip:bob@biloxi.com>;tag=456248 Call-ID: 843817637684230@998dasdh09 CSeq: 1826 REGISTER Contact: <sip:bob@192.0.2.4> Expires: 7200 Content-Length: 0 |
| address of record whose registration is to be created, queried, or modified. The To header field and the Request-URI field typically differ, as the former contains a user name. This address-of-record MUST be a SIP URI or SIPS URI. From: The From header field contains the address-of-record of the person responsible for the registration. The value is the same as the To header field unless the request is a third-party registration. Call-ID: All registrations from a UAC SHOULD use the same Call-ID header field value for registrations sent to a particular registrar. If the same client were to use different Call-ID values, a registrar could not detect whether a delayed REGISTER request might have arrived out of order. CSeq: The CSeq value guarantees proper ordering of | |
| Call-ID: 843817637684230@998sdasdh09 CSeq: 1826 REGISTER Contact: <sip:bob@192.0.2.4> Expires: 7200 Content-Length: 0 | |
| | | | |
| REGISTER requests. A UA MUST increment the CSeq value by one for each REGISTER request with the same Call-ID. Contact: REGISTER requests MAY contain a Contact header field with zero or more values containing address bindings. |
|---|---|---|---|
| section 10.2 page 58 | 148 | We can write Test case for UAC. | As per rfc3261 UAs MUST NOT send a new registration (that is, containing new Contact header field values, as opposed to a retransmission) until they have received a final response from the registrar for the previous one or the previous REGISTER request has timed out. UAC MUST NOT send a new registration until they have received a final response from the registrar for the previous one |
| 72 | UAs MUST NOT send a new registration (that is, containing new Contact header field values, as opposed to a retransmission) until they have received a final response from the registrar for the previous one or the previous REGISTER request has timed out. F1: UAC->UAS |
| REGISTER sip:registrar.biloxi.com SIP/2.0 Via:SIP/2.0/UDPbobspc.biloxi.com:5060;branch=z9hG4bKnashds7 Max-Forwards: 70 To: Bob <sip:bob@biloxi.com> From: Bob <sip:bob@biloxi.com>;tag=456248 Call-ID: 843817637684230@998sdasdh09 CSeq: 1826 REGISTER Contact: <sip:bob@192.0.2.4> Expires: 7200 Content-Length: 0 |
| F2: UAS->UAC |
| SIP/2.0 200 ok Via: SIP/2.0/UDP bobspc.biloxi.com:5060;branch=z9hG4bKnashds7;received=192.0.2.4 To: Bob <sip:bob@biloxi.com>;tag=2493k59kd From: Bob <sip:bob@biloxi.com>;tag=456248 Call-ID: 843817637684230@998sdasdh09 CSeq: 1826 REGISTER Contact: <sip:bob@192.0.2.4> Expires: 7200 Content-Length: 0 |
| 73 | If the address-of-record in the To header field of a REGISTER request is a SIPS URI, then any Contact header field |
| section 10.2 page 58 | This section is covered in test case 146 |

If the address-of-record in the To header field of a REGISTER request is a SIPS URI, then any Contact header field
| values in the request SHOULD also be SIPS URIs. | section 11.2 page 68 | 149 | We as UAC check the presence of these Header field in response for a OPTIONS request. | UAS | As per RFC 3261, Allow, Accept, Accept-Encoding, Accept-Language, and Supported header fields SHOULD be present in a 200 (OK) response to an OPTIONS request. Here, we as UAC sends an OPTIONS request to UAS. UAS SHOULD include Accept, accept-encoding, accept-Language in response. | UAS SHOULD include Accept, accept-encoding, accept-Language in response. | F1: UAC->UAS
OPTION sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

F2: UAC->UAS
SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Contact: <sip:alice@pc33.atlanta.com>
Accept-Language: da, en-gb;q=0.8, en;q=0.7
Accept-Encoding: gzip
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0

F1: UAC->UAS
OPTION sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

F2: UAC->UAS
SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Contact: <sip:alice@pc33.atlanta.com>
Accept-Language: da, en-gb;q=0.8, en;q=0.7
Accept-Encoding: gzip
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
<table>
<thead>
<tr>
<th>75</th>
<th>The UAS MUST add a Contact header field to the response.</th>
<th>section 12.1.1 page 70</th>
<th>150</th>
<th>We as UAC check the presence of Contact Header filed in response.</th>
<th>UAS</th>
<th>As per rfc3261 the UAS MUST add a Contact header field to the response. Here we as UAC sends an INVITE to UAS. UAS MUST include Contact header field in response.</th>
<th>UAS MUST include Contact header field in 200 ok response.</th>
<th>F1: UAC-&gt;UAS</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE Max-Forwards: 70 Contact: <a href="">sip:alice@pc33.atlanta.com</a> Content-Length: 568</td>
<td></td>
<td></td>
</tr>
<tr>
<td>76</td>
<td>A UAS MUST be prepared to receive a request without a tag in the From field, in which</td>
<td>section 12.1.1 page 71</td>
<td>151</td>
<td>For s.no 76 and 112 we UAC check whether the UAS accept the</td>
<td>UAS</td>
<td>As per rfc3261 a UAS MUST be prepared to receive a request without a</td>
<td>UAS MUST accept the INVITE and response back</td>
<td>F1: UAC-&gt;UAS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>77</td>
<td>If the request has a Request-URI or a topmost Route header field value with a SIPS URI, the Contact header field MUST contain a SIPS URI.</td>
<td>This section is covered in test case 119</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>78</td>
<td>A UAC MUST be</td>
<td>section 12.1.2 page 71</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

If the request has a Request-URI or a topmost Route header field value with a SIPS URI, the Contact header field MUST contain a SIPS URI. This section is covered in test case 119.
prepared to receive a response without a tag in the To field, in which case the tag is considered to have a value of null.

12.1.2 page 72

whether UAC accept the response without To tag or not.

whether UAC is prepared to receive a response without a tag in the To field, in which case the tag is considered to have a value of null. Here we as UAS sends response to UAC but without To tag. UAC MUST be prepared to receive it.

UAC MUST be prepared to receive a response without a tag in the To field, in which case the tag is considered to have a value of null.

UAS chosen its value

72 The Call-ID of the request MUST be set to the Call-ID of the dialog.

section 12.1.2 page 73

This section is covered in test case 137

80 if the local sequence number is not empty,

section 12.1.2 page 153

For this section we write two test

UAS as per rfc3261 if the local sequence

UAS MUST chosen its value

F1: UAC->UAS

INVITE sip: bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS

ACK sip: bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
the value of the local sequence number MUST be incremented by one, and this value MUST be placed into the CSeq header field. If the local sequence number is empty, an initial value MUST be chosen using the guidelines of Section 8.1.1.5.

<table>
<thead>
<tr>
<th>Cases</th>
<th>73</th>
<th>Number is not empty, the value of the local sequence number MUST be incremented by one, and this value MUST be placed into the CSeq header field. If the local sequence number is empty, an initial value MUST be chosen using the guidelines of Section 8.1.1.5.</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. To check the behavior of UAS on empty Cseq field.</td>
<td></td>
<td></td>
</tr>
<tr>
<td>2. To check whether the UAC increase the Cseq no on new request or not.</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>as per guideline of rfc3261</th>
<th>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</th>
</tr>
</thead>
<tbody>
<tr>
<td>Via: SIP/2.0/UDP</td>
<td></td>
</tr>
<tr>
<td>pc33.atlanta.com;branch=z9hG4bKnashds8</td>
<td></td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
<td></td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td></td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
<td></td>
</tr>
<tr>
<td>CSeq: 314159</td>
<td></td>
</tr>
<tr>
<td>Max-Forwards: 70</td>
<td></td>
</tr>
<tr>
<td>Contact: <a href="">sip:alice@pc33.atlanta.com</a></td>
<td></td>
</tr>
<tr>
<td>Content-Length: 568</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>F2: UAS-&gt;UAC</th>
<th>SIP/2.0 200 ok</th>
</tr>
</thead>
<tbody>
<tr>
<td>Via:SIP/2.0/UDP</td>
<td></td>
</tr>
<tr>
<td>pc33.atlanta.com;branch=z9hG4bKnashds8</td>
<td></td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
<td></td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td></td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
<td></td>
</tr>
<tr>
<td>CSeq: 314159</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>F3: UAC-&gt;UAS</th>
<th>ACK sip:bob@192.0.2.4 SIP/2.0</th>
</tr>
</thead>
<tbody>
<tr>
<td>Via: SIP/2.0/UDP</td>
<td></td>
</tr>
<tr>
<td>pc33.atlanta.com;branch=z9hG4bKnashds8</td>
<td></td>
</tr>
<tr>
<td>Max-Forwards: 70</td>
<td></td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
<td></td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td></td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
<td></td>
</tr>
<tr>
<td>CSeq: 314159</td>
<td></td>
</tr>
</tbody>
</table>

UAC in the second case we as UAS check whether the UAC increase the cseq no on new request or not. UAC send INVITE to UAS. We as UAS send 200 ok. On 200 ok UAC MUST increase the Cseq no.

<table>
<thead>
<tr>
<th>F1: UAC-&gt;UAS</th>
<th>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</th>
</tr>
</thead>
<tbody>
<tr>
<td>Via: SIP/2.0/UDP</td>
<td></td>
</tr>
<tr>
<td>pc33.atlanta.com;branch=z9hG4bKnashds8</td>
<td></td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
<td></td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
<td></td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
<td></td>
</tr>
<tr>
<td>CSeq: 314159</td>
<td></td>
</tr>
</tbody>
</table>

73
we as UAS ask UAC to send re-INVITE message and check whether the cseq no increases or not. UAC MUST increase the Cseq no.

Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC
SIP/2.0  200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS
ACK sip:bob@biloxi.com SIP/2.0
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK

F4: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq:314160
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

F5: UAS->UAC
SIP/2.0  200 ok
Via:SIP/2.0/UDP
81 If the route set is empty, the UAC MUST place the remote target URI into the Request-URI. The UAC MUST NOT add a Route header field to the request.

It is not possible to write test case for it. Reason being how we can expect as UAS that UAC will have empty root header filed

82 If the request has a tag in the To header field, but the dialog identifier does not match any existing dialogs, the UAS may have crashed and restarted, or it may have received a request for a different (possibly failed) UAS (the UASs can construct the To tags so that a UAS can identify that the tag was for a UAS for which it is providing recovery). Another possibility is that the incoming request has been simply misrouted. Based on the To tag, the UAS MAY either accept or reject the request. Accepting the request for acceptable

Here we are going to Test the Behavior of UAS on receiving a request with To tag. In SFTF case the request with the To tag is rejected with 481.

Here we are going to Test the Behavior of UAS on receiving a request with To tag. We as UAC sends an INVITE message with To tag in it. UAS reject the request.

UAS reject the INVITE with 481.

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 481 Call transction does not exist
To tags provides robustness, so that dialogs can persist even through crashes. UAs wishing to support this capability must take into consideration some issues such as choosing monotonically increasing CSeq sequence numbers even across reboots, reconstructing the route set, and accepting out-of-range RTP timestamps and sequence numbers.

| 83 | If the UAS wishes to reject the request because it does not wish to recreate the dialog, it MUST respond to the request with a 481 (Call/Transaction Does Not Exist) status code and pass that to the server transaction. | section 12.2.2 page 76 | This section is covered in test case 145 |

| 84 | It is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. The UAS MUST then set the remote sequence number to the value of the sequence number. | section 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | 12.2.2 page 77 | as per rfc3261 it is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | 12.2.2 page 77 | UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | 12.2.2 page 77 | F1: UAC->UAS |

| 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS |

| 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS |

| 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS | as per rfc3261 it is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | UAS | SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | F1: UAC->UAS | INVITE sip:bob@biloxi.com SIP/2.0 |

| 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS | as per rfc3261 it is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | UAS | SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | F1: UAC->UAS |

| 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS | as per rfc3261 it is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | UAS | SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | F1: UAC->UAS | INVITE sip:bob@biloxi.com SIP/2.0 |

| 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS | as per rfc3261 it is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | UAS | SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | F1: UAC->UAS |

| 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS | as per rfc3261 it is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | UAS | SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | F1: UAC->UAS |

| 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS | as per rfc3261 it is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | UAS | SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | F1: UAC->UAS | INVITE sip:bob@biloxi.com SIP/2.0 |

| 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS | as per rfc3261 it is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | UAS | SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | F1: UAC->UAS |

| 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS | as per rfc3261 it is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | UAS | SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | F1: UAC->UAS | INVITE sip:bob@biloxi.com SIP/2.0 |

| 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS | as per rfc3261 it is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | UAS | SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | F1: UAC->UAS |

| 12.2.2 page 77 | 156 | We as UAC send a request with increase value in cseq more than one and check the behavior of UAS. | UAS | as per rfc3261 it is possible for the CSeq sequence number to be higher than the remote sequence number by more than one. This is not an error condition, and a UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | UAS | SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request. | F1: UAC->UAS | INVITE sip:bob@biloxi.com SIP/2.0 |
| 85 | The UAC MAY add an Expires header field (Section 20.19) to limit the | section 13.2.1 page 79 | 157 | As per rfc3261The UAC MAY add an Expires header field (Section | UAC | As per rfc3261The UAC MAY add an Expires header field (Section | UAC core SHOULD generate a CANCEL | F1: UAC->UAS |
| F1: UAC->UAS | INVITE sip:bob@biloxi.com SIP/2.0 | Via: SIP/2.0/UDP | pc33.atlanta.com;branch=z9hG4bKnashds8 | To: Bob <sip:bob@biloxi.com>;tag=a6c85cf | From: Alice <sip:alice@atlanta.com>;tag=1928301774 | Call-ID: a84b4c76e66710 | CSeq: 314159 INVITE |
| F2: UAS->UAC | sip:bob@192.0.2.4 SIP/2.0 | Via: SIP/2.0/UDP | pc33.atlanta.com;branch=z9hG4bKnashds8 | To: Bob <sip:bob@biloxi.com>;tag=a6c85cf | From: Alice <sip:alice@atlanta.com>;tag=1928301774 | Call-ID: a84b4c76e66710 | CSeq: 314161 BYE |
| | | | | | | | Content-Length: 0 |

The UAC MUST then set the remote sequence number to the value of the sequence number in the CSeq header field value in the request. Here we as UAC sends an INVITE message to UAS. On 200 ok response we sends BYE request to UAS with CSeq sequence number to be higher than the remote sequence number by more than one. UAS SHOULD be prepared to receive and process requests with CSeq values more than one higher than the previous received request.

m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

As per rfc3261

The UAC MAY add an Expires header field (Section 20.19) to limit the

UAC core SHOULD generate a CANCEL

F1: UAC->UAS INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP
The initial offer MUST be in either an INVITE or, if not there, in the first reliable non-failure message from the UAS back to the UAC. In this specification, that is the final 2xx response.

| validity of the invitation.  If the time indicated in the Expires header field is reached and no final answer for the INVITE has been received, the UAC core SHOULD generate a CANCEL request for the INVITE, as per Section 9. | 20.19) to limit the validity of the invitation. If the time indicated in the Expires header field is reached and no final answer for the INVITE has been received, the UAC core SHOULD generate a CANCEL request for the INVITE, as per Section 9. Here, UAC sends INVITE with Expiry Header Field. We as UAS does not sends any response to UAC. UAC core SHOULD generate a CANCEL request for the INVITE. | 20.19) to limit the validity of the invitation. If the time indicated in the Expires header field is reached and no final answer for the INVITE has been received, the UAC core SHOULD generate a CANCEL request for the INVITE, as per Section 9. Here, UAC sends INVITE with Expiry Header Field. We as UAS does not sends any response to UAC. UAC core SHOULD generate a CANCEL request for the INVITE. | 20.19) to limit the validity of the invitation. If the time indicated in the Expires header field is reached and no final answer for the INVITE has been received, the UAC core SHOULD generate a CANCEL request for the INVITE, as per Section 9. Here, UAC sends INVITE with Expiry Header Field. We as UAS does not sends any response to UAC. UAC core SHOULD generate a CANCEL request for the INVITE. | 20.19) to limit the validity of the invitation. If the time indicated in the Expires header field is reached and no final answer for the INVITE has been received, the UAC core SHOULD generate a CANCEL request for the INVITE, as per Section 9. Here, UAC sends INVITE with Expiry Header Field. We as UAS does not sends any response to UAC. UAC core SHOULD generate a CANCEL request for the INVITE. | pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:bob@biloxi.com> From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 Cseq: 314159 expire: 60 Max-Forwards: 70 Contact: <sip:alice@pc33.atlanta.com> Content-Length: 568 v=0 o=UAC 2890844526 2890844526 IN IP4 here.com s=Session SDP c=IN IP4 pc33.atlanta.com t=0 0 m=audio 49172 RTP/AVP 0 a=rtpmap:0 PCMU/8000 F2: UAC->UAS CANCEL sip:bob@192.0.2.4 SIP/2.0 Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:bob@biloxi.com> From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 Cseq: 314159 INVITE F3: UAS->UAC SIP/2.0 200 ok Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 CANCEL |

| 86 The initial offer MUST be in either an INVITE or, if not there, in the first reliable non-failure message from the UAS back to the UAC. In this specification, that is the final 2xx response. | 158 13.2.1 page 79 | 86 The initial offer MUST be in either an INVITE or, if not there, in the first reliable non-failure message from the UAS back to the UAC. In this specification, that is the final 2xx response. | 158 13.2.1 page 79 |

As per rfc3261 if the INVITE does not contain a session description, the UAS is being asked to participate in a session, and the UAC has asked that the UAS MUST contain an offer in 200 ok response.

<p>| 86 The initial offer MUST be in either an INVITE or, if not there, in the first reliable non-failure message from the UAS back to the UAC. In this specification, that is the final 2xx response. | 158 13.2.1 page 79 | 86 The initial offer MUST be in either an INVITE or, if not there, in the first reliable non-failure message from the UAS back to the UAC. In this specification, that is the final 2xx response. | 158 13.2.1 page 79 | 86 The initial offer MUST be in either an INVITE or, if not there, in the first reliable non-failure message from the UAS back to the UAC. In this specification, that is the final 2xx response. | 158 13.2.1 page 79 | F1: UAC-&gt;UAS INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq:314159 Max-Forwards: 70 Contact: <a href="">sip:alice@pc33.atlanta.com</a> |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th>provide the offer of the session. It MUST provide the offer in its first non-failure reliable message back to the UAC. In this specification, that is a 2xx response to the INVITE. Here we as UAC sends an INVITE message to UAS without Session description. UAS MUST contain an offer in 200 ok response</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>F2: UAS-&gt;UAC</td>
<td>SIP/2.0 200 ok</td>
<td>Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE Content-Length: 568</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>v=0 o=UAS 2890844526 2890844526 IN IP4 here.com s=Session SDP c=IN IP4 pc33.atlanta.com t=0 0 m=audio 49172 RTP/AVP 0 a=rtpmap:0 PCMU/8000</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>F3: UAC-&gt;UAS</td>
<td>ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 Max-Forwards: 70 To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 ACK Content-Length: 568</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>v=0 o=UAS 2890844526 2890844526 IN IP4 here.com s=Session SDP c=IN IP4 pc33.atlanta.com t=0 0 m=audio 49172 RTP/AVP 0 a=rtpmap:0 PCMU/8000</td>
<td></td>
<td></td>
</tr>
<tr>
<td>87</td>
<td>If the initial offer is in an INVITE, the answer MUST be in a reliable non-failure message from UAS back to UAC which is correlated to that INVITE. For this specification, that is only the final 2xx section 13.2.1 page 80</td>
<td>We can write Test case for UAC.</td>
<td>UAC As per rfc3261 UAC MUST treat the first session description it receives as the answer, and MUST ignore any session descriptions in subsequent UAC MUST treat the first session description it receives as the answer, and MUST ignore any session descriptions in</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>F1: UAC-&gt;UAS</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
| response to that INVITE. That same exact answer MAY also be placed in any provisional responses sent prior to the answer. The UAC MUST treat the first session description it receives as the answer, and MUST ignore any session descriptions in subsequent responses to the initial INVITE. | responses to the initial INVITE. Here we as UAS sends two different SDP with two 200 ok response with session descriptions. | subsequent responses to the initial INVITE. | Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
| | | | Content-Length: 568
v=0
o=UAS 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
F2: UAS->UAC
SIP/2.0 200 ok
Via:SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
| Content-Length: 568
v=0
o=UAS 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
F2: UAS->UAC
SIP/2.0 200 ok
Via:SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
| Content-Length: 569
v=0
o=UAS 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49188 RTP/AVP 0
a=rtpmap:0 PCMU/8000
F2: UAS->UAC
SIP/2.0 200 ok
Via:SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
| Content-Length: 569
v=0
o=UAS 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49188 RTP/AVP 0
a=rtpmap:0 PCMU/8000

<table>
<thead>
<tr>
<th>Section</th>
<th>Page 80</th>
<th>Line 160</th>
</tr>
</thead>
<tbody>
<tr>
<td>88</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

If the initial offer is in the first reliable non-failure message from the UAS back to UAC, the answer MUST be in the acknowledgment for that message (in this specification, ACK for a 2xx response).

We as UAS check the behavior of UAC on not receiving offer in 200 ok.

As per rfc3261 if the initial offer is in the first reliable non-failure message from the UAS back to UAC, the answer MUST be in the acknowledgment for that message (in this specification, ACK for a 2xx response). Here we are going opposite to it. We as UAC sends INVITE without session description. UAS sends session description with 200 ok. On sending ACK we do not send session description. UAS MUST response back with 4xx.

UAS MUST response back with 415.

---

F4: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK

F1: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>

F2: UAS->UAC
SIP/2.0 200 ok
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Length: 0

F3: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP
pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK

F4: UAS->UAC
SIP/2.0 415
Via: SIP/2.0/UDP
<table>
<thead>
<tr>
<th>Section</th>
<th>Page</th>
<th>Line</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>13.2.1</td>
<td>80</td>
<td>89</td>
<td>Once the UAS has sent or received an answer to the initial offer, it MUST NOT generate subsequent offers in any responses to the initial INVITE. This means that a UAS based on this specification alone can never generate subsequent offers until completion of the initial transaction.</td>
</tr>
<tr>
<td>13.2.1</td>
<td>80</td>
<td>161</td>
<td>We can write Test case for UAS.</td>
</tr>
<tr>
<td>13.2.1</td>
<td>80</td>
<td></td>
<td>UAS As per rfc3261 once the UAS has sent or received an answer to the initial offer, it MUST NOT generate subsequent offers in any responses to the initial INVITE. This means that a UAS based on this specification alone can never generate subsequent offers until completion of the initial transaction. Here we as UAC sends INVITE to UAS. On multiple same response we check whether UAS sends multiple offers in it or not. UAS MUST NOT generate subsequent offers in any responses to the initial INVITE.</td>
</tr>
<tr>
<td>13.2.1</td>
<td>80</td>
<td></td>
<td>UAS MUST NOT generate subsequent offers in any responses to the initial INVITE.</td>
</tr>
</tbody>
</table>

F3: UAC->UAS

ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq:314159
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC

SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Content-Length: 568
v=0
<table>
<thead>
<tr>
<th>Section</th>
<th>Description</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>90</td>
<td>Once the INVITE has been passed to the INVITE client transaction, the UAC waits for responses for the INVITE. If the INVITE client transaction returns a timeout rather than a response the TU acts as if a 408 (Request Timeout) response had been received, as described in Section 8.1.3.</td>
<td>This cannot be whitebox tested</td>
</tr>
<tr>
<td>91</td>
<td>A 3xx response may contain one or more Contact header field values providing new addresses where the callee might be reachable. Depending on the status code of the 3xx response (see Section 21.3), the UAC MAY choose to try those new addresses.</td>
<td>Same as case 123</td>
</tr>
<tr>
<td>92</td>
<td>The UAC core MUST generate an ACK</td>
<td>We can write Test case for UAC.</td>
</tr>
</tbody>
</table>
If the request is an "2xx" response from the transaction layer, the header fields of the ACK are constructed in the same way as for any request sent within a dialog (see Section 12) with the exception of the CSeq and the header fields related to authentication. The sequence number of the CSeq header field MUST be the same as the INVITE being acknowledged, but the CSeq method MUST be ACK. The ACK MUST contain the same credentials as the INVITE. If the "2xx" contains an offer (based on the rules above), the ACK MUST carry an answer in its body. If the offer in the "2xx" response is not acceptable, the UAC core MUST generate a valid answer in the ACK and then send a BYE immediately.

<table>
<thead>
<tr>
<th>page</th>
<th>same CSeq no. as for original request and if 200 ok response contain offer than UAC MUST sends offer in ACK. Here we as UAS sends 200 ok with offer for INVITE sends by UAC. On ACK we as UAS check whether ACK have same CSeq and a offer.</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>CSeq no. in ACK as for original request and if 200 ok response contain offer than UAC MUST sends offer in ACK.</td>
</tr>
</tbody>
</table>

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c7e6e6710
CSeq: 314159 INVITE
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>

ACK  sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c7e6e6710
CSeq: 314159 ACK
Content-Length: 568
v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F3: UAC->UAS

ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c7e6e6710
CSeq: 314159 ACK
Content-Length: 568
v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F1: UAC->UAS
<table>
<thead>
<tr>
<th>13.3.1 page 83</th>
<th>but for UAS</th>
<th>the request is an INVITE that contains an Expires header field, the UAS core sets a timer for the number of seconds indicated in the header field value. When the timer fires, the invitation is considered to be expired. If the invitation expires before the UAS has generated a final response, a 487 (Request Terminated) response SHOULD be generated.</th>
<th>response back with 487.</th>
<th>13.3.1 page 83</th>
<th>164</th>
<th>This section is covered in test case 154</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE that contains an Expires header field, the UAS core sets a timer for the number of seconds indicated in the header field value. When the timer fires, the invitation is considered to be expired. If the invitation expires before the UAS has generated a final response, a 487 (Request Terminated) response SHOULD be generated.</td>
<td>but for UAS</td>
<td>the request is an INVITE that contains an Expires header field, the UAS core sets a timer for the number of seconds indicated in the header field value. When the timer fires, the invitation is considered to be expired. If the invitation expires before the UAS has generated a final response, a 487 (Request Terminated) response SHOULD be generated. Here we as UAC sends INVITE to UAS with Expire header field value zero. UAS MUST response back with 487.</td>
<td>response back with 487.</td>
<td>13.3.1 page 83</td>
<td>164</td>
<td>This section is covered in test case 154</td>
</tr>
<tr>
<td>Section</td>
<td>Content</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>---------</td>
<td>---------</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>12.2.2</td>
<td>for a different (possibly failed) UAS. Section 12.2.2 provides guidelines to achieve a robust behavior under such a situation.</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>13.3.1</td>
<td>If the INVITE does not contain a session description, the UAS is being asked to participate in a session, and the UAC has asked that the UAS provide the offer of the session. It MUST provide the offer in its first non-failure reliable message back to the UAC. In this specification, that is a 2xx response to the INVITE. This section is covered in test case 158.</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>13.3.1.3</td>
<td>A common scenario occurs when the callee is currently not willing or able to take additional calls at this end system. A 486 (Busy Here) SHOULD be returned in such a scenario. We can write Test case for UAS to check its behavior when UAS busy.</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>13.3.1.3</td>
<td>A common scenario occurs when the callee is currently not willing or able to take additional calls at this end system. A 486 (Busy Here) SHOULD be returned in such a scenario. Here we as UAC sends INVITE to UAS. After ACK response we sends another INVITE with different value. UAS is already engage with first INVITE and SHOULD response to second INVITE with 486.</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Example INVITE message:**

F1: UAC->UAS  
INVITE sip:bob@biloxi.com SIP/2.0  
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8  
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf  
From: Alice <sip:alice@atlanta.com>;tag=1928301774  
Call-ID: a84b4c76e66710  
Cseq: 314159  
Max-Forwards: 70  
Contact: <sip:alice@pc33.atlanta.com>  
Content-Length: 568  
v=0  
c=Session SDP  
s=IN IP4 here.com  
t=0  
m=audio 49172 RTP/AVP 0  
a=rtpmap:0 PCMU/8000  
F2: UAS->UAC  
SIP/2.0 180 ringing  
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8  
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
<table>
<thead>
<tr>
<th>Section</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>13.3.1.4</td>
<td>If the INVITE request contained an offer, and the UAS had not yet sent an answer, the 2xx MUST contain an answer. If the INVITE did not contain an offer, the 2xx MUST contain an offer if the UAS had not yet sent an offer.</td>
</tr>
</tbody>
</table>

This section is covered in test case 158.
<p>|   | UAC MAY send a re-INVITE with no session description, in which case the first reliable non-failure response to the re-INVITE will contain the offer (in this specification, that is a 2xx response). | section 14.1 page 86 | 166 | This case is same as SDP in INVITE but here the case is re-INVITE we check the behavior of UAS on session modification. | UAS | As per rfc3261 UAC MAY send a re-INVITE with no session description, in which case the first reliable non-failure response to the re-INVITE will contain the offer (in this specification, that is a 2xx response). In this case we as UAC sends an INVITE message to UAS. On 200 ok response from UAS we send re-INVITE message to UAS with no session description. UAS MUST response back with 200 ok contain session description. | F1: UAC-&gt;UAS | UAS MUST response back with 200 ok contain session description. |
|---|---|---|---|---|---|---|---|
| F1: UAC-&gt;UAS | INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 1 Max-Forwards: 70 Contact: <a href="">sip:alice@pc33.atlanta.com</a> |
| F2: UAS-&gt;UAC | SIP/2.0 200 ok Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 1 INVITE Content-Length: 568 |
| F3: UAC-&gt;UAS | ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 Max-Forwards: 70 To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 1 ACK Content-Length: 568 |
| F4: UAC-&gt;UAS | INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24k1id85 |</p>
<table>
<thead>
<tr>
<th>99</th>
<th>If there is an ongoing INVITE client transaction, the TU MUST wait until the transaction reaches the completed or terminated state before initiating the new INVITE.</th>
<th>section 14.1 page 87</th>
<th>168</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>As per rfc3261f there is an ongoing INVITE client transaction, the TU MUST wait until the transaction reaches the completed or terminated state before initiating the new INVITE. Here</td>
<td>UAS</td>
<td></td>
</tr>
<tr>
<td></td>
<td>As per rfc3261f there is an ongoing INVITE client transaction, the TU MUST wait until the transaction reaches the completed or terminated state before initiating the new INVITE. Here</td>
<td>UAC wait until transaction reaches the completed or terminated state before initiating the new INVITE. Here</td>
<td>F1: UAC-&gt;UAS</td>
</tr>
<tr>
<td></td>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</td>
<td>Via: SIP/2.0/UDP</td>
<td>pc33.atlanta.com;branch=z9hG4bKnashds8</td>
</tr>
<tr>
<td></td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Call-ID: a84b4c76e66710</td>
<td>CSeq: 1</td>
<td>Max-Forwards: 70</td>
</tr>
<tr>
<td></td>
<td>Content-Length: 568</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>ACK sip:bob@192.0.2.4 SIP/2.0</td>
<td>Via: SIP/2.0/UDP</td>
<td>pc33.atlanta.com;branch=z9hG4b4b24klid85</td>
</tr>
<tr>
<td></td>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
<td>From: Alice <a href="">sip:alice@atlanta.com</a></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Call-ID: a84b4c76e66710</td>
<td>CSeq: 2 ACK</td>
<td>Content-Length: 568</td>
</tr>
</tbody>
</table>
UAC sends INVITE to UAS. We as UAC check whether UAC sends INVITE during ongoing transaction of first INVITE. UAC wait until transaction reaches the completed or terminated state before initiating the new INVITE.

UAC sends INVITE to UAS. We as UAS check whether UAC sends second INVITE during on going transaction of first INVITE. UAC wait until transaction reaches the completed or terminated state before initiating the new INVITE.

Content-Length: 568
v=0
o=UAC 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC
SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 1 INVITE

F3: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 1ACK

If there is an ongoing INVITE server transaction, the TU MUST wait until the transaction reaches the confirmed or terminated state before initiating the new INVITE.

This section is covered in test case 169

If a UA receives a non-2xx final response to a re-INVITE, the session parameters MUST remain unchanged, as if no re-INVITE had been issued. Note that, as stated in Section 12.2.1.2, if the non-2xx

We can write a test case to check the behavior of UAC on session modification.

As per rfc3261 If a UA receives a non-2xx final response to a re-INVITE, the session parameters MUST remain unchanged, as if no re-INVITE had been.

UAC MUST terminate the modified session and return to older one.

F1: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 1
<table>
<thead>
<tr>
<th>F1: UAC-&gt;UAS</th>
<th>F2: UAS-&gt;UAC</th>
</tr>
</thead>
<tbody>
<tr>
<td>final response is a 481 (Call/Transaction Does Not Exist), or a 408(Request Timeout), or no response at all is received for the re-INVITE (that is, a timeout is returned by the INVITE client transaction), the UAC will terminate the dialog.</td>
<td>issued. Note that, as stated in Section 12.2.1.2, if the non-2xx final response is a 481 (Call/Transaction Does Not Exist), or a 408 (Request Timeout), or no response at all is received for the re-INVITE (that is, a timeout is returned by the INVITE client transaction), the UAC will terminate the dialog. Here, UAC sends INVITE message to UAS. We as UAS sends 200 ok. On 200 ok UAC sends re-INVITE. On receiving re-INVITE we as UAS sends 481 response UAC MUST terminate the session. Also UAC MUST not terminate the first INVITE session.</td>
</tr>
</tbody>
</table>
| Max-Forwards: 70  
Contact: <sip:alice@pc33.atlanta.com>  
Content-Length: 568  
v=0  
o=UAS 2890844526 2890844526 IN IP4 here.com  
s=Session SDP  
c=IN IP4 pc33.atlanta.com  
t=0 0  
m=audio 49172 RTP/AVP 0  
a=rtpmap:0 PCMU/8000  
F2: UAS->UAC | Max-Forwards: 70  
Contact: <sip:alice@pc33.atlanta.com>  
Content-Length: 568  
v=0  
o=UAS 2890844526 2890844526 IN IP4 here.com  
s=Session SDP  
c=IN IP4 pc33.atlanta.com  
t=0 0  
m=audio 49172 RTP/AVP 0  
a=rtpmap:0 PCMU/8000  
F2: UAS->UAC |
A UAS that receives a second INVITE before it sends the final response to a first INVITE with a lower Cseq sequence number on the same dialog MUST return a 500 (Server Internal Error) response to the second INVITE.

<table>
<thead>
<tr>
<th>Section</th>
<th>Page</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>14.2</td>
<td>88</td>
<td>We can write a test case to check the behavior of UAS on receiving a request with lower Cseq no than the previous request.</td>
</tr>
</tbody>
</table>

**UAS**

As per rfc3261 a UAS that receives a second INVITE before it sends the final response to a first INVITE with a lower Cseq sequence number on the same dialog MUST return a 500 (Server Internal Error) response to the second INVITE. Here we as UAC sends INVITE to UAS. On 180 ringing response from UAS we as UAC sends another INVITE with lower Cseq no. UAS return a 500 (Server Internal Error) response to the second INVITE.

**UAS return a 500 (Server Internal Error) response to the second INVITE.**

<table>
<thead>
<tr>
<th>Sequence</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>F1: UAC-&gt;UAS</td>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnasbd8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a> Call-ID: a84b4c76e66710 CSeq: 2 Content-Length: 568</td>
</tr>
<tr>
<td>F2: UAS-&gt;UAC</td>
<td>SIP/2.0 180 ringing Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnasbd8 To: Bob <a href="">sip:bob@biloxi.com</a> From: Alice <a href="">sip:alice@atlanta.com</a> Call-ID: a84b4c76e66710</td>
</tr>
</tbody>
</table>

---

92

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<p>| 103 | If the new session description is not acceptable, the UAS can reject it by returning a 488 (Not Acceptable Here) response for the re-INVITE. | section 14.2 page 89 | 173 | we as UAC check the behavior of UAS on receiving a method in re-invite which UAS not support. | UAS | As per rfc3261 if the new session description is not acceptable, the UAS can reject it by returning a 488 (Not Acceptable Here) response for the re-INVITE. Here we as UAC sends INVITE to UAS. on 200 ok we sends re-INVITE to UAS will response back with 488 and continue with old session. | F1: UAC→UAS |
| CSeq:2 INVITE | F3: UAC→UAS |
| INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 | INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0 |
| Via: SIP/2.0/UDP | Via: SIP/2.0/UDP |
| pc33.atlanta.com;branch=z9hG4b24klid85 | pc33.atlanta.com;branch=z9hG4b24klid85 |
| To: Bob <a href="">sip:bob@biloxi.com</a> | To: Bob <a href="">sip:bob@biloxi.com</a> |
| From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928302564 | From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928302564 |
| Call-ID: a84b4c76e66710 | Call-ID: a84b4c76e66710 |
| CSeq: 1 | CSeq: 1 |
| Max-Forwards: 70 | Max-Forwards: 70 |
| Contact: <a href="">sip:alice@pc33.atlanta.com</a> | Contact: <a href="">sip:alice@pc33.atlanta.com</a> |
| Content-Length: 560 | Content-Length: 560 |
| F4: UAS→UAC | F5: UAC→UAS |
| SIP/2.0 500 (Server Internal Error) | ACK sip:bob@192.0.2.4 SIP/2.0 |
| Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24klid85 | Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24klid85 |
| To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf | To: Bob <a href="">sip:bob@biloxi.com</a>;tag=1928302564 |
| From: Alice <a href="">sip:alice@atlanta.com</a>;tag=a6c85cf | From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928302564 |
| Call-ID: a84b4c76e66710 | Call-ID: a84b4c76e66710 |
| CSeq: 1 INVITE | CSeq: 1 ACK |</p>
<table>
<thead>
<tr>
<th>Step</th>
<th>Message</th>
<th>Details</th>
</tr>
</thead>
</table>
| F2: | UAS→UAC | SIP/2.0 200 ok
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 1 INVITE
Content-Length: 568 |
| F3: | UAC→UAS | ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 568 |
| F4: | UAC→UAS | INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24klid85
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>
Call-ID: a84b4c76e66710
CSeq: 2
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 560 |
| F5: | UAS→UAC | SIP/2.0 488
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24klid85

UAS, with session description that UAS does not support. UAS will response back with 488
| 104 | If a UAS generates a 2xx response and never receives an ACK, it SHOULD generate a BYE to terminate the dialog. | section 14.2 page 89 | 174 | We can write one single Test to test the behavior of UAS on not receiving ACK | UAS | As per rfc3261 if a UAS generates a 2xx response and never receives an ACK, it SHOULD generate a BYE to terminate the dialog. Here We as UAC sends INVITE message to UAS. On receiving 200 ok response we as UAC not send ACK. UAS SHOULD generate a BYE to terminate the session. | UAS | SHOULD generate a BYE to terminate the session. |

To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928302564
Call-ID: a84b4c76e66710
CSeq: 2

F6: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24kld85
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928302564
Call-ID: a84b4c76e66710
CSeq: 2

F1: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Cseq: 314159
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC
SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159

F3: UAS->UAC

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| Page 105 | Section 15.1.2 page 91 | 175 | UAS response back with 481. | UAS | As per rfc3261 a UAS core receiving a BYE request checks if it matches an existing dialog. If the BYE does not match an existing dialog, the UAS core SHOULD generate a 481 (Call/Transaction Does Not Exist) response and pass that to the server transaction. Here we as UAC sends an INVITE to UAS. On receiving 200 ok response we as UAC sends BYE request that doesn't match to the current session. UAS response back with 481.

A UAS core receiving a BYE request checks if it matches an existing dialog. If the BYE does not match an existing dialog, the UAS core SHOULD generate a 481 (Call/Transaction Does Not Exist) response and pass that to the server transaction.

F1: UAC -> UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F2: UAS -> UAC

SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F4: UAC -> UAS

SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

BYE sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160 BYE

F3: UAS -> UAC

SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568
v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F4: UAC -> UAS

SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568
v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
<table>
<thead>
<tr>
<th>F3: UAC-&gt;UAS</th>
</tr>
</thead>
<tbody>
<tr>
<td>BYE sip:bob@192.0.2.4 SIP/2.0</td>
</tr>
<tr>
<td>Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8</td>
</tr>
<tr>
<td>Max-Forwards: 70</td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td>Call-ID: b2765dhdjadh2w23</td>
</tr>
<tr>
<td>CSeq: 3141 BYE</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>F4: UAS-&gt;UAC</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP/2.0 481 Call transaction does not exist</td>
</tr>
<tr>
<td>Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8</td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td>Call-ID: b276dhdjadhw223</td>
</tr>
<tr>
<td>CSeq: 3141 BYE</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>F5: UAS-&gt;UAC</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP/2.0 200 ok</td>
</tr>
<tr>
<td>Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8</td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a>;tag=a6c85cf</td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td>Call-ID: b2765dhdjadhw223</td>
</tr>
<tr>
<td>CSeq: 3141 BYE</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>106 The CSeq header field in the ACK MUST contain the same value for the sequence number as was present in the original request, but the method parameter MUST be equal to &quot;ACK&quot;.</th>
</tr>
</thead>
<tbody>
<tr>
<td>section 17.1.1.3 page 129</td>
</tr>
<tr>
<td>176 Same as case 116 but for ACK method here we check whether the ACK use same Cseq no or not.</td>
</tr>
<tr>
<td>UAC This test is same as case 116 but here we are checking whether the UAC use same cseq field value as in original request. UAC send a request to UAS. We as UAS sends 200ok. On 200 ok response UAC sends ACK, we as UAS check whether UAC use same Cseq no as in original request.</td>
</tr>
<tr>
<td>UAC MUST include same Cseq no in ACK as in original request.</td>
</tr>
<tr>
<td>176 UAC-&gt;UAS</td>
</tr>
<tr>
<td>INVITE sip:<a href="mailto:bob@biloxi.com">bob@biloxi.com</a> SIP/2.0</td>
</tr>
<tr>
<td>Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8</td>
</tr>
<tr>
<td>To: Bob <a href="">sip:bob@biloxi.com</a></td>
</tr>
<tr>
<td>From: Alice <a href="">sip:alice@atlanta.com</a>;tag=1928301774</td>
</tr>
<tr>
<td>Call-ID: a84b4c76e66710</td>
</tr>
<tr>
<td>Cseq: 314159</td>
</tr>
<tr>
<td>Max-Forwards: 70</td>
</tr>
<tr>
<td>Contact: <a href="">sip:alice@pc33.atlanta.com</a></td>
</tr>
<tr>
<td>Content-Length: 568</td>
</tr>
<tr>
<td>CSeq: 314159</td>
</tr>
<tr>
<td>Max-Forwards: 70</td>
</tr>
<tr>
<td>Contact: <a href="">sip:alice@pc33.atlanta.com</a></td>
</tr>
<tr>
<td>Content-Length: 568</td>
</tr>
<tr>
<td>v=0 o=UA 2890844526 2890844526 IN IP4 here.com s=Session SDP</td>
</tr>
<tr>
<td>Case</td>
</tr>
<tr>
<td>------</td>
</tr>
<tr>
<td>107</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>108</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>109</td>
</tr>
<tr>
<td></td>
</tr>
</tbody>
</table>

UAC MUST include same Cseq no and Method in ACK as in original request.

```
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

F2: UAS->UAC

```
SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c75e6e6710
CSeq: 314159 INVITE
```

F3: UAS->UAC

```
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c75e6e6710
CSeq: 314159 ACK
```
oriented transports such as TCP, the Content-Length header field indicates the size of the body. The Content-Length header field MUST be used with stream oriented transports.

| 110 | any given parameter-name MUST NOT appear more than once. | section 19.1.1 page 149 | Same as case 106 |
| 111 | URIs MUST NOT contain unescaped space and control characters. | section 19.1.1 page 152 | Same as case 102 |
| 112 | When a UA sends a request outside of a dialog, it contains a From tag only, providing “half” of the dialog ID. | section 19.3 page 159 | This section is covered in test case 151 |
| 113 | When a tag is generated by a UA for insertion into a request or response, it MUST be globally unique and cryptographically random with at least 32 bits of randomness. A property of this selection requirement is that a UA will place a different tag into the From header of an INVITE than it would place into the To header of the response to the same INVITE. | section 19.3 page 159 | 178 Here we as UAC check whether the To tag and From tag in response is different or same. |

UAS

As per rfc3261 When a tag is generated by a UA for insertion into a request or response, it MUST be globally unique and cryptographically random with at least 32 bits of randomness. A property of this selection requirement is that a UA will place a different tag into the From header of an INVITE than it would place into the To header of the response to the same INVITE.

UAS MUST put different value in To and From Tag.

F1: UAC->UAS

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Cseq: 314159
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC
| 114 | When present in an INVITE request, the Alert-Info header field specifies an alternative ring tone to the UAS. When present in a 180 (Ringing) response, the Alert-Info header field specifies an alternative ringback tone to the UAC. | section 20.4 page 164 | 179 | We can write a Test case as UAS | UAS |

In this case we as UACS sends INVITE containing a Alter-Info specifies an alternative ring tone to the UAS. On 180 ringing the Alert-Info header field specifies an alternative ring back tone to the UAC. |

On 180 ringing the Alert-Info header field specifies an alternative ring back tone to the UAC. | F1: UAC->UAS |

INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c7e66710
CSeq: 314159 ACK |

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtmap:0 PCMU/8000 |

F2: UAS->UAC |

SIP/2.0 180 ringing
Alert-Info: <http://www.example.com/sounds/moo.wav>
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774 |
<table>
<thead>
<tr>
<th>Sequence</th>
<th>Description</th>
<th>Page</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>115</td>
<td>The Content-Encoding header field is used as a modifier to the &quot;media-type&quot;. When present, its value indicates what additional content codings have been applied to the entity-body, and thus what decoding mechanisms MUST be applied in order to obtain the media-type referenced by the Content-Type header field.</td>
<td>20.12</td>
<td>This section is covered in test case 110</td>
</tr>
<tr>
<td>116</td>
<td>If a stream-based protocol (such as TCP) is used as transport, the Content-Length header field MUST be used.</td>
<td>20.14</td>
<td>Same as case 111</td>
</tr>
<tr>
<td>117</td>
<td>If no body is present in a message, then the Content-Length header field value MUST be set to zero.</td>
<td>20.14</td>
<td>This section is covered in test case 111</td>
</tr>
<tr>
<td>118</td>
<td>The Retry-After header field can be used with a 500 (Server Internal Error) or 503 (Service Unavailable) response to indicate how long the</td>
<td>20.33</td>
<td>We can write one test for UAC</td>
</tr>
</tbody>
</table>

Unavailability response to indicate how long the service is expected to be unavailable to the requesting client and with a 404 (Not Found), 413 (Request Entity Too Large), 480 (Temporarily Unavailable), 486 (Busy Here), 600 (Busy), or 603 (Decline) response to indicate when the called party anticipates being available again.

The value of this field is a positive integer number of seconds (in decimal) after the time of the response. An optional "duration" parameter indicates how long the called party will be reachable starting at the initial time of availability. If no duration parameter is given, the service is assumed to be available indefinitely.

Examples:

```
Retry-After: 18000;duration=3600
Retry-After: 120 (I'm in a meeting)
```

From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Cseq: 314159
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568

v=0
o=UserName 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F2: UAS->UAC
SIP/2.0 486
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Retry-After: 18000;duration=3600
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F3: UAC->UAS
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK

F4: UAC->UAS
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160
Max-Forwards: 70
Contact: <sip:alice@pc33.atlanta.com>
Content-Length: 568
<p>| | | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>119</td>
<td>The Request-URI was ambiguous. The response MAY contain a listing of possible unambiguous addresses in Contact header fields. Revealing alternatives can infringe on privacy of the user or the organization. It MUST be possible to configure a server to respond with status 404 (Not Found) or to suppress the listing of possible choices for</td>
<td>182</td>
<td>As per rfc3261 the response to INVITE MAY contain a listing of possible unambiguous addresses in Contact header fields. Here We as UAS sends 485 Ambiguous for UAC INVITE. UAC use one of the URI in Contact Header Field in response.</td>
<td>UAC</td>
</tr>
</tbody>
</table>

---

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 pc33.atlanta.com
f=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

F5: UAS->UAC
SIP/2.0 200 ok
Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE

F6: UAS->UAC
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314160 ACK
<table>
<thead>
<tr>
<th>ambiguous Request-URIs.</th>
<th>Example response to a request with the Request-URI sip:<a href="mailto:lee@example.com">lee@example.com</a>:</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>SIP/2.0 485 Ambiguous Contact: Carol Lee <a href="">sip:carol.lee@example.com</a></td>
</tr>
<tr>
<td></td>
<td>Contact: Ping Lee <a href="">sip:p.lee@example.com</a> Contact: Lee M. Foote <a href="">sips:lee.foote@example.com</a></td>
</tr>
<tr>
<td></td>
<td>Some email and voice mail systems provide this functionality. A status code separate from 3xx is used since the semantics are different: for 300, it is assumed that the same person or service will be reached by the choices provided. While an automated choice or sequential search makes sense for a 3xx response, user intervention is required for a 485 (Ambiguous) response.</td>
</tr>
</tbody>
</table>

| From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |
| From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
| From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
| From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
| From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |
| From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
| From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
| From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
| From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |

From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 INVITE |
From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
From: Alice <sip:alice@atlanta.com>;tag=1928301774 Contact: Carol Lee <sip:carol.lee@example.com> Contact: Ping Lee <sip:p.lee@example.com> Contact: Lee M. Foote <sips:lee.foote@example.com> Call-ID: a84b4c76e66710 CSeq: 314159 ACK |
### 4.0 RFC 3261 TEST CASE EXCEPTIONS

The following test cases could not be tested using SFTF due to the following reasons:

<table>
<thead>
<tr>
<th>S.no</th>
<th>RFC3261 Section</th>
<th>Source</th>
<th>Reason</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>section 7 page 28</td>
<td>This case works fine if we as UAS check the presence of single space (SP) characters, but we as UAC cannot send INVITE without single space (SP) REQUEST as the SipRequest code in SFTF is hard coded as shown below: self.event.headers.insert(0, self.method + &quot; &quot; + self.rUri.create() + &quot; &quot; + self.protocol + &quot;/&quot;&quot; + self.version + &quot;\r\n&quot;)</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>section 13.2.2 page 81</td>
<td>It is not possible to write a test case for this since we as UAS cannot know that UAC treated a time out error as 408.</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>section 8.1.3.1 page 42</td>
<td>Same as above</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>section 8.1.3.1 page 42</td>
<td>It is not possible to write test case for this as we as UAS cannot know that UAC treat a fatal transport error as 503.</td>
<td></td>
</tr>
</tbody>
</table>

---

These test cases demonstrate the limitations of SFTF in testing certain scenarios as described in RFC 3261. It highlights the need for more flexible testing frameworks that can handle such edge cases better.