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RFC3261 UAC/UAS Test Cases

For

Sip Forum Test Frame Work (SFTF)

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Revision History

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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

#	RFC 3261 excerpt	RFC location	Test Case	Can this be combined with another clause?	Under Test	Test Case Description	Expected Behaviour	Call Flow
1	messages consist of a start-line, one or more header fields, an empty line indicating the end of the header fields.	section 7 page 27	101	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 send by UAC and in Second way test the behavior of UAS on missing one of these line.	UAS	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.	UAS will response back with 400 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/2.0 400 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: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
2	start-line, each message-header line, and the empty line MUST be terminated by a carriage-return line-feed sequence (CRLF)	section 7 page 27		This section is covered in test case 101	UAC	In the second case we as UAS check the INVITE send by UAC is as per rfc3261 or not.	UAC will have to send INVITE as per rfc3261 format.	F1: UAC->UAS INVITE sip:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8

3	Note that the empty line MUST be present even if the message-body is not.	section 7 page 27		This section is covered in test case 101				
4	A Request-Line contains a method name, a Request-URI, and the protocol version separated by a single	section 7 page 28	102	For s.no. 4, 5 and 111 we can write one single test case in two way in first case we as	UAC	As per rfc3261 the Rrequest 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			UAS check the whether the Request line and Request-URI in Invite of UAC is as per rfc3261 or not.		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.		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/2.0 603 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
5	The Request-URI MUST NOT contain unescaped spaces or control characters	section 7 page 28		This section is covered in test case 102	UAS	In the Second case we as UAC send request without Method and containing escaped space in URI. UAS will reject the Request with 400.	UAS will response back with 400 Bad request.	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/2.0 400 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 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
6	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.	section 7 page 28	103	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 Sio-Version.	UAC	As per rfc3261 all the request and response MUST contain SIP-Version as "SIP/2.0". In this case we as UAS check the presence of SIP-version in INVITE send by UAC.	UAC MUST include SIP-Version in the 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 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 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: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 F4: UAS->UAC 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 Content-Length: 0 F5: UAC->UAS SIP/2.0 200 OK Via: SIP/2.0/UDP 192.0.2.4;branch=z9hG4bKnashds10 From: Bob <sip:bob@biloxi.com>;tag=a6c85cf To: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 Cseq:314160 BYE Content-Length: 0

					UAS	In the Third case we as UAC sends INVITE message to UAS with lower SIP-Version UAS MUST accept the INVITE	UAS MUST accept the 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 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
7	A Status-Line consists	section 7	104	For s.no 7,8 and 9	UAS	As per rfc3261 the	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 formate.		response is differ fron 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.	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 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
8	No CR or LF is allowed except in the final CRLF sequence.	section 7 page 28		This section is covered in test case 104	UAC	In the Second case we as UAS send response with out	UAC will not understand response and	F1: UAC->UAS INVITE sip:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP

						status line. UAC will not understand response and discard the message.	discard the message.	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 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: 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 Contact: <sip:bob@192.0.2.4> 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

9	Status-Line = SIP-Version SP Status-Code SP Reason-Phrase CRLF	section 7 page 28	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 formate or not by sending Invite with this specific format.		For this muti 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	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=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 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
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 "field-name: field-value" pair, without changing the semantics of the message, by appending each subsequentfield-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					

	line are treated as a single SP character							
16	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.	section 7.3.1 page 31	This section is covered in test case 105					
17	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.	section 7.3.1 page 32	106	Here we as UAC check the behavior of UAS on repeated parameter name of Header Field..	UAS	As per rfc3261 arbitrary number of parameter pairs may be attached to a header field value, any given parameter-name MUST NOT appear more than once. Here we as UAC sends INVITE to UAS with repeated parameter name.	UAS will response back with 400 bad reques 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> Contact: <sip:alice@pc33.atlanta.com> 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/2.0 400 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: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
18	When comparing header fields, field names are always case- insensitive.	section 7.3.1 page 32	107	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.		Here we sends an INVITE message with header field name random case(lower and upper mix). UAS MUST accept this format.	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> 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 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
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					
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	108	Here we check the behavior of UAC on getting a request with response.		As per rfc3261if 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 sends a request with response. UAC MUST ignore it.	UAC MUST ignore the Request Header Field.	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 200ok 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>;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
21	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.	section 7.3.3 page 33	109	Here we check whether the UAS support l the header field name in both short and long form or not.	UAS	As per rfc3261field 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.	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 t: Bob<sip:bob@biloxi.com> To: Bob<sip:bob@biloxi.com> f: Alice <sip:alice@atlanta.com>;tag=1928301774 From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 i: a84b4c76e66710 CSeq: 314159 INVITE Max-Forwards: 70 Contact:<sip:alice@pc33.atlanta.com> m: <sip:alice@pc33.atlanta.com> 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

								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
22	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.	section 7.4.1 page 33	110	For s.no 22 and 115 we as UAC check the behavior on missing one Content-Encoding field while sending a encoded message.	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.	UAS will not understand coding and response back 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 Date: Thu, 21 Feb 2002 13:02:03 GMT Contact: <sip:alice@pc33.atlanta.com> Content-Length: 568 v=# o=%!@# s=***** c=^>$$@ t=------ m=nnff a=rtpmap:0 PCMU/8000 F2: UAS->UAC

								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 reciving INVITE without coded body but with Content-Encoding Field.	UAS wil 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

								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
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,	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> 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/TCP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf

						In the second case we as UAC send INVITE with body but with content – length value zero	UAS reject the INVITE with 415	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/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 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> 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: 0 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/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: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/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
24	All SIP implementations MUST support the SIP URI scheme.	section 8.1.1.2 page 36	This section is covered in test case 105					
25	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.	section 8.1.1.2 page 36	112	Here we as UAS check the presence of To-Tag(UAC must not add To tag in Invite)	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

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

								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
26	A UAC SHOULD use the display name "Anonymous", along with a syntactically correct, but otherwise meaningless URI (like sip:thisis@anonymous.invalid), if the identity of the client is to remain hidden.	section 8.1.1.3 page 37	113	We can write Test case for UAS.	UAS	As per rfc3261 UAC SHOULD use the display name "Anonymous", along with a syntactically correct, but otherwise meaningless URI (like sip:thisis@anonymous.invalid), if the identity of the client is to remain hidden. here we as UAC sends INVITE with display name Anonymous if the identity of the client is to remain hidden. UAS response back with 108 ringing/ 200 ok.	UAS response back with 180 ringing/ 200 ok.	F1: UAC->UAS INVITE sip:thisis@anonymous.invalid SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob<:thisis@anonymous.invalid>;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 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
27	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.	section 8.1.1.4 page 37	114	For s.no 27, 29 and 31 we check whether the Call-ID is same throughout Session. this test can be done in two way. 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.On200 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.	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 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: 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
					UAC	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	UAC MUST maintain same Call-ID Field value throughout 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

						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 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@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

								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
28	Call-ID SHOULD be the same in each registration from a UA.	section 8.1.1.4 page 37	115	Same as case115 but for Registration method. S..no. 28 and 30 are combine in this test.	UAC	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.	UAC MUST use same Call-ID field value in second registration request.	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 401Authentication requir 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 F3: 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: 1827 REGISTER Contact: <sip:bob@192.0.2.4> Expires: 7200

								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
29	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.	section 8.1.1.4 page 37	This section is covered in test case 114					
30	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	section 8.1.1.4 page 38	This section is covered in test case 115					
31	Call-IDs are case-sensitive and are simply compared byte-by-byte.	section 8.1.1.4 page 38	This section is covered in test case 114					
32	The CSeq header field serves as a way to identify and order transactions. It consists	section 8.1.1.5 page 38	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	F1: UAC->UAS INVITE sip:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP

	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.	responce.	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 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
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	117	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.	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

						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 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 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
34	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	118	For s.no 34 and 35 we can write one Test case in two way one as UAC and check whether the response send by UAS contain Via and second one as UAS to check whether UAC Invite contain Via or not.	UAC	As per rfc3261 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. UAC sends INVITE to UAS. We as UAS check the presence of Via	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

						header field in it, the SIP- Version and the branch parameter		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/2.0 603 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
35	The Via header field value MUST contain a branch parameter.	section 8.1.1.7 page 38	This section is covered in test case 118					
36	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	section 8.1.1.8 page 40	119	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.	UAC	As per rfc3261 if the request contain SIP-URI, Contact header field MUST contain a SIPS URI as well and MUST be present in all request. Here we as UAS check the INVITE whether INVITE sends by UAC Contact Header field or not and contain SIP-URI. UAC MUST contain Contact	UAC MUST contain Contact Header field and SIP-URI in it.	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

	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 time out 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.					

	connection failures in TCP), the condition MUST be treated as a 503 (Service Unavailable) status code.							
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 reconise and treat it as 400 response code.	UAC does not reconise 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=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/2.0 432 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

								Content-Length: 0
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 reconise and treat it as 183 and wait for 200 ok.	UAC does not reconise 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

								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
42	If more than one Via header field value is present in a response, the UAC SHOULD discard the message.	section 8.1.3.3 page 43	122	We can write Test case for UAC.	UAC	As per rfc3261if 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.	UAC should consider that response is miss routed 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=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 200ok Via:SIP/2.0/UDP pc33.atlanta.com;pc35.altlanta.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>;tag=1928301774 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,>

								From: Alice<sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE 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
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.	section 8.1.3.4 page 43	123	For s.no 43,44,45,46and 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	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.	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/2.0 301 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf Contact: <sip: unknown@biloxi.com: ;,hari@biloxi.com > From: Alice<sipsmith@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314159 INVITE

						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 t=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:hari@biloxi.com>;tag=a6c85cf From: Alice<sipsmith@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314160 INVITE F6: UAC->UAS ACK sip:hari@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4babcd1234 Max-Forwards: 70 To: Bob <sip:hari@biloxi.com>;tag=a6c85cf From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710

								CSeq: 314160 ACK Content-Length: 0
44	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.	section 8.1.3.4 page 43	This section is covered in test case 124					
45	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.	section 8.1.3.4 page 43	This section is covered in test case 124					
46	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.	section 8.1.3.4 page 44	This section is covered in test case 124					
47	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	section 8.2.1 page 46	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.	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->UAS UNKNOWN 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

	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 as UAC sends method which UAS does not support. UAS MUST response back with 405 also add Contact Header Field which showers the method Which UAS support.	support.	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/2.0 405 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
48	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.	section 8.2.2 page 46	126	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.	UAS	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	UAS MUST ignore this header field and response back with 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> From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 Cseq: 314159 INVITE Max-Forwards: 70 ignore: mustingnore@ignore.com Contact: <sip:alice@pc33.atlanta.com> Content-Length: 568

						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 define in a specification and not necessary for the processing of INVITE. UAS MUST ignore this header field and response back with 200 ok.		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@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
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 rfc3261it 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	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>;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

	response with a 403 (Forbidden) status code and pass it to the server transaction for transmission.					user of this UAS. Here we as UAC sends INVITE to UAS containing URI scheme (for example, a tel: URI) in the To header field which UAS do not recognize. UAS accept requests even if they do not recognize the URI scheme		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/2.0 403 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	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	UAS MUST response back with 482(loop dected)	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, 00 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

						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 dected)		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 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 dected 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	As per rfc3261 if same request has arrived at the UAS more than once,	UAS MUST response back with 482(loop ducted)	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 message to UAS with different path in Via header field. UAS MUST response back with 482(loop dected)		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=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/2.0 180 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 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 dected 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
53	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.		131	We can write Test case for UAS.	UAS	As per rfc3261the 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	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	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: UAS->UAC SIP/2.0 420 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 Usupported: 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

								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
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 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/2.0 108 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/2.0 200 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
				UAC	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.	UAC MUST not include these Header Field.	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 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 CANCEL Content-Length: 0
					UAS	In the third case we as UAC send INVITE to UAS.On 180 ringing we send CANCEL request containing Require header fileld. UAS MUST ignore this header filed present in the request	UAS MUST ignore this header filed present in the request	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/2.0 180 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 Require: Pyhton 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 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
					UAS	In the fourth case we as UAC sends INVITE to UAS. On 200 ok response we sends ACK containing Require header fileld. UAS MUST ignore these header fileld.	UAS MUST ignore these header fileld.;	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 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 Require: new method 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
55	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.	section 8.2.3 page 48	133	We can write Test case for UAS.	UAS	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	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.	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-Encoding: unknown Content-Language: xyz 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/2.0 415 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

						(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
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

								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
57	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.	section 8.2.6.1 page 49	135	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 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 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@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
58	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 response and receipt of the request, measured in seconds.	section 8.2.6.2 page 50	136	We can write Test case for UAS.	uas	As per rfc3261When 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 response 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.	UAS SHOULD add a delay value into the Timestamp value in the 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 Timestamp : 54 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 Timestamp:54 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
59	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.	section 8.2.6.2 page 50	137	For s.no. 59 and 79 We can write a single Test which can check the whether all these three field have same value or not.	UAS	As per rfc3261the 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. Here we as UAC sends an INVITE to UAS. On the response from UAS we check whether the From header field and Call-ID header field have the same field value or not. UAS MUST use same Field value in response.	UAS MUST use same Field value in 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 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/UDPpc33.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
60	If a request contained a To tag in the request, the To header field in	section 8.2.6.2 page 50	138	We can write Test case for UAS.	UAS	As per rfc3261 if a request contained a To tag in the	UAS MUST contain same To-Tag.	F1: UAC->UAS INVITE sip:bob@biloxi.com SIP/2.0

	the response MUST equal that of the request.					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 conatains same To tag or not. UAS MUST contain same To-Tag.		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 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

								CSeq: 314160 BYE Content-Length: 0 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: 314160 BYE
61	if the To header field in the request did not contain a tag, the URI in the Toheader 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	section 8.2.6.2 page 50	139	We can write Test case for UAS to check whether it accept a request without To tag and add To tag in response.	UAS	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	UAS on response MUST add to tag in 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 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@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> From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314160 BYE Content-Length: 0 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: 314160 BYE
62	In that usage, a UAS that receives a CANCEL request for an INVITE, but has not yet sent a final response, would "stop ringing", and then respond to the INVITE with a specific error response (a 487).	section 9 page 53	140	For s.no 62 and 69 we can write Test case for UAS to test its behavior on receiving Cancel request.	UAS	As per rfc3261 in that usage, a UAS that receives a CANCEL request for an INVITE, but has not yet sent a final response, would "stop ringing", and then respond to the INVITE with a specific error response (a 487). Here we as UAC sends an INVITE to UAS. On receiving 180ringing we sends a CANCEL request to UAS.	UAS would "stop ringing", and then respond to the INVITE with a specific error response (a 487).	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

						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
63	If the transaction for the original request still exists, the behavior of the UAS on receiving a	section 9.2 page 55	141	For s.no .63 and 64 we write can write one single test for UAS	UAS	As per rfc3261 iif the transaction for the original request still exists, the	CANCEL request has no effect on the processing of	F1: UAC->UAS INVITE sip:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP

	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.					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. here we as UAC sends INVITE message to UAS. On 200 ok we send CANCEL request to UAS. CANCEL request has no effect on the processing of the original request	the original request	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> 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/2.0 400 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

								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
64	A CANCEL request has no impact on the processing of transactions with any other method defined in this specification.	section 9.2 page 55	This section is covered in test case 141					
65	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.	section 9.2 page 55	142	We can write Test case for UAS.	UAS	As per rfc3261regardless 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. Here we as UAC sends INVITE message to UAS. On 180 ringing we as UAC sends CANCEL request to UAS. UAS MUST respond with 200ok.	UAS MUST respond with 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/2.0180 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/2.0 200 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
66	The CANCEL request MUST NOT contain any Require or Proxy-Require header fields.	section 9.1 page 54	143	we can write a test case to check the presence of Require in cancel request send by UAC.	UAC	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.	The request MUST NOT contain any Require or Proxy-Require header fields.	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. 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/2.0 200 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	144	For s.no 68 and 83 we as UAC check the Behavior of UAS on miss match cancel request	UAS	As per rfc3261The 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	The destination address, port, and transport for the CANCEL MUST be identical to those used to send the original request.	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 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

						INVITE message to UAS. We as UAS sends 180 ringing. UAC sends a CANCEL request. We as UAS check whether the UAC use same port and transport or not.		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 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
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	section 9.2 page 55	145	For s.no 68 and 83 we as UAC check the Behavior of UAS on miss match cancel request	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

	(Call Leg/Transaction Does Not Exist).					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 200ok 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.		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: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 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
69	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).	section 9.2 page 55	This section is covered in test case 140					
70	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.	Section 10.2 page 55	146	For s.no .70 and 73 we write can write one single test for UAC	UAC	Asper rfc3261 Record-Route header field has no meaning in REGISTER requests or responses, and MUST be ignored if present and 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. Here UAC sends REGISTER request to UAS. We as UAS check the presence of RECORD-Route header field and Contact header	UAC MUST NOT include Record-Route header field. and MUST contain SIP URI in Contact header field.	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

						field SIP URI.		
					UAS	In the second case we as UAC sends Register request with Record-Route header Field. UAS MUST ignore the Record- route Header filed.	UAS MUST ignore the Record- route Header filed.	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 Record-Route: <sip:ss2.biloxi.example.com;lr> 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
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 location 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.	UAC	As per rfc3261 REGISTER Request MUST include Request-URI, To, Call-ID, Cseq,Contact Header fields. We as UAS check the presence of these filed in Register request sends by UAC. UAC MUST include all these header fields in Request.	UAC MUST include all these header fields in Request.	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

	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.							
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.	section 10.2 page 58	148	We can write Test case for UAC.	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. Here we as UAS check whether the UAC sends REGISTER request more than once before giving final responce.	UAC MUST NOT send a new registration until they have received a final response from the registrar for the previous one	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					

	values in the request SHOULD also be SIPS URIs.							
74	Allow, Accept, Accept-Encoding, Accept-Language, and Supported header fields SHOULD be present in a 200 (OK) response to an OPTIONS request.	section 11.2 page 68	149	We as UAC check the presence of these Header field in response for a OPTION request.	UAS	As per rfc3261Allow, 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 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 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

75	The UAS MUST add a Contact header field to the response.	section 12.1.1 page 70	150	We as UAC check the presence of Contact Header filed in response.	UAS	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	UAS MUST include Contact header field in 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 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 Contact: <sip:alice@pc33.atlanta.com> 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
76	A UAS MUST be prepared to receive a request without a tag in the From field, in which	section 12.1.1 page 71	151	For s.no 76 and 112 we UAC check whether the UAS accept the	UAS	As per rfc3261 a UAS MUST be prepared to receive a request without a	UAS MUST accept the INVITE and response back	F1: UAC->UAS INVITE sip:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP

	case the tag is considered to have a value of null.			INVITE without From tag.		tag in the From field, in which case the tag is considered to have a value of null. Here we as UAC sends an INVITE to UAS without From tag. UAS MUST accept the INVITE and response back with 200 ok.	with 200 ok.	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> 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 Contact: <sip:alice@pc33.atlanta.com> 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
77	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.	section 12.1.2 page 71	This section is covered in test case 119					
78	A UAC MUST be	section	152	We as UAS Check	UAC	As per rfc3261 a	UAC MUST be	F1: UAC->UAS

	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.		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. Here we as UAS sends response to UAC but without To tag. UAC MUST prepare to receive it.	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	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 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
79	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

	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.	73		cases 1. To check the behavior of UAS on empty Cseq field. 2. To check whether the UAC increase the Cseq no on new request or not.		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. Here in first case we as UAC sends INVITE message to UAS with empty Cseq header field value. UAS MUST chosen its value.	as per guideline of rfc3261	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: 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@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
					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.	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

						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

								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 INVITE F6: 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: 314160 ACK Content-Length: 0
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.	section 12.1.2 page 74	It is not possible to write test case for it. Reason being how we can expect as UAS that UAC will have empty root heaer fileld					
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	section 12.2.2 page 76	154	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.	UAS	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.							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
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.	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 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

	in the CSeq header field value in the request.					The UAS 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 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 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: 314161 BYE Content-Length: 0 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: 314161
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 INVITE sip:bob@biloxi.com SIP/2.0 Via: SIP/2.0/UDP

	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 Expire 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 Expire Header Field. We as UAS does not sends any response to UAC. UAC core SHOULD generate a CANCEL request for the INVITE	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=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->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.	section 13.2.1 page 79	158	For s.no .86, 95 and 97 we write can write one single test to check the Behavior of UAS on not receiving the offer in invite.	UAS	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	UAS MUST contain an offer in 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 Max-Forwards: 70 Contact: <sip:alice@pc33.atlanta.com>

						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		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=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: a84b4c76e66710 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
87	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	159	We can write Test case for UAC.	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	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

	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=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 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: 569 v=0 o=UserA 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

								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
88	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).	section 13.2.1 page 80	160	We as UAS check the behavior of UAC on not receiving offer in 200ok.	UAC	As per rfc326If 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.	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

								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: 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
89	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.	section 13.2.1 page 80	161	We can write Test case for UAS.	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.	UAS MUST NOT generate subsequent offers in any responses to the initial 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 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

								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: a84b4c76e66710 CSeq: 314159 ACK
90	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.	section 13.2.2 page 81	This cannot be whitebox tested					
91	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.	section 13.2.2.2 page 81	Same as case 123					
92	The UAC core MUST generate an ACK	section 13.2.2.4	162	We can write Test case for UAC.	UAC	As per rfc3261 the ACK have the	UAC MUST have the same	F1: UAC->UAS

	request for each 2xx received 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 Section12) 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.	page 82				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.	Cseq no. in ACK as for original request and if 200 ok response contain offer than UAC MUST sends offer in ACK.	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: 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: 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: a84b4c76e66710 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
93	If the request is an	section	163	Same as case 160	UAS	As per rfc3261 If	UAS MUST	F1: UAC->UAS

	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.	13.3.1 page 83		but for UAS		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.	response back with 487.	 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 expire: 0 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/2.0 487 Request terminated 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 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
94	If the request has a tag in the To header field but the dialog identifier does not match any of the existing dialogs, the UAS may have crashed and restarted, or may have received a request	section 13.3.1 page 83	164	This section is covered in test case 154				

	for a different(possibly failed) UAS. Section 12.2.2 provides guidelines to achieve a robust behavior under such a situation.							
95	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.	section 13.3.1 page 84	This section is covered in test case 158					
96	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.	section 13.3.1.3 page 85	165	we can write Test case for UAS to check its behavior when UAS busy.	UAS	As per rfc3261 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.	UAS is already engage with first INVITE and SHOULD response to second INVITE with 486.	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/2.0 180 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 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:314160 Max-Forwards: 70 Contact: <sip:alice@pc33.atlanta.com> Content-Length: 568 F4: UAS->UAC SIP/2.0 486 Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:bob@biloxi.com>;tag=a6c85ht From: Alice<sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 314160 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
97	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.	section 13.3.1.4 page 85	This section is covered in test case 158						

98	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 contains session description.	UAS MUST response back with 200 ok contain session description.	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 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: 1 INVITE 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: a84b4c76e66710 CSeq: 1 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>;tag=a6c85cf From: Alice <sip:alice@atlanta.com>;tag=1928302564 Call-ID: a84b4c76e66710 CSeq: 2 Max-Forwards: 70 Contact: <sip:alice@pc33.atlanta.com> F5: UAC->UAS SIP/2.0 200 ok Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24klid85 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf From: Alice<sip:alice@atlanta.com>;tag=1928302564 Call-ID: a84b4c76e66710 CSeq: 2 INVITE 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 F6: UAC->UAS ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24klid85 Max-Forwards: 70 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq: 2 ACK Content-Length: 568
99	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.	section 14.1 page 87	168	As per rfc3261if 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	UAS	As per rfc3261if 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	UAC wait until transaction reaches the completed or terminated state before initiating the new 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: 1 Max-Forwards: 70 Contact: <sip:alice@pc33.atlanta.com>

				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 send ssecond 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=UserA 2890844526 2890844526 IN IP4 here.com s=Sescond 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
100	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.	section 14.1 page 87	169	This section is covered in test case 169				
101	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	section 14.1 page 87	170	We can write a test case to check the behavior of UAC on session modification.	UAC	As per rfc3261If 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

	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.					issued.Note that, as stated in Section 12.2.1.2, if the non-2xx finalresponse 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 200ok 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.		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: 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>;tag=1928302564 Call-ID: a84b4c76e66710 CSeq: 2 Max-Forwards: 70 Contact: <sip:alice@pc33.atlanta.com> Content-Length: 560 F5: UAS->UAC

								SIP/2.0 481 Call transaction does not exist Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24klid85 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf From: Alice<sip:alice@atlanta.com>;tag=1928302564 Call-ID: a84b4c76e66710 CSeq: 1 INVITE F6: UAC->UAS ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24klid85 Max-Forwards: 70 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 CSeq:2 ACK
102	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	section 14.2 page 88	172	We can write a test case to check the behavior of UAS on receiving a request with lower Cseq no than the previous request.	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.	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: 2 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/2.0 180 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:2 INVITE F3: 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>;tag=1928302564 Call-ID: a84b4c76e66710 CSeq: 1 Max-Forwards: 70 Contact: <sip:alice@pc33.atlanta.com> Content-Length: 560 F4: UAS->UAC SIP/2.0 500 (Server Internal Error) Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24klid85 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf From: Alice<sip:alice@atlanta.com>;tag=1928302564 Call-ID: a84b4c76e66710 CSeq: 1 INVITE F5: UAC->UAS ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24klid85 Max-Forwards: 70 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf From: Alice <sip:alice@atlanta.com>;tag=1928302564 Call-ID: a84b4c76e66710 CSeq:1 ACK
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 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 Max-Forwards: 70 Contact: <sip:alice@pc33.atlanta.com> Content-Length: 568

						UAS. with session description that UAS does not support. UAS will response back with 488		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: 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>;tag=1928302564 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

								To: Bob <sip:bob@biloxi.com>;tag=a6c85cf From: Alice<sip:alice@atlanta.com>;tag=1928302564 Call-ID: a84b4c76e66710 CSeq: 2 INVITE F6: UAC->UAS ACK sip:bob@192.0.2.4 SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4b24klid85 Max-Forwards: 70 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf From: Alice <sip:alice@atlanta.com>;tag=1928302564 Call-ID: a84b4c76e66710 CSeq:2 ACK
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 200ok 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.	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 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: UAS->UAC

								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 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: 314160 BYE
105	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.	section 15.1.2 page 91	175	Same as case 147 but the method is BYE	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.	UAS 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> 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

								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>;tag=a6c85cf From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: b276dhdjfhdw223 CSeq: 3141 BYE F4: UAS->UAC SIP/2.0 481 Call transaction does not exist 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: b276dhdjfhdw223 CSeq: 3141 BYE 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: b276dhdjfhdw223 Cseq: 3141 BYE
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 "ACK".	section 17.1.1.3 page 129	176	Same as case 116 but for ACK method here we check whether the ACK use same Cseq no or not.	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.	UAC MUST include same Cseq no in ACK as in original request.	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 Content-Length: 568 v=0 o=UserA 2890844526 2890844526 IN IP4 here.com s=Session SDP

						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: a84b4c76e66710 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: a84b4c76e66710 CSeq: 314159 ACK
107	If the response has the same value of the branch parameter in the top Via header field as the branch parameter in the top Via header field of the request that created the transaction.	section 17.1.3 page 132	Same as case 115 the purpose of test is to check whether the same header field parameter is use throughout the session or not.					
108	If the method parameter in the CSeq header field matches the method of the request that created the transaction. The method is needed since a CANCEL request constitutes a different transaction, but shares the same value of the branch parameter.	section 17.1.3 page 132	Same as case 116					
109	In the case of stream-	section 18.3	Same as case 111					

	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.	page 147						
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

						Here We as UAC sends INVITE message to UAS. On 200 ok response we as UAC check whether the To and From tag are same or not. UAS MUST put different value in To and From Tag and they MUST be unique.		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
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 UAC 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> From: Alice <sip:alice@atlanta.com>;tag=1928301774 Call-ID: a84b4c76e66710 Alert-Info: <http://www.example.com/sounds/moo.wav> 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/2.0 180 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

								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
115	The Content-Encoding header field is used as a modifier to the "media-type". 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.	section 20.12 page 169	This section is covered in test case 110					
116	If a stream-based protocol (such as TCP) is used as transport, the Content Length header field MUST be used.	section 20.14 page 169	Same as case 111					
117	If no body is present in a message, then the Content-Length header field value MUST be set to zero.	section 20.14 page 169	180	This section is covered in test case 111				
118	The Retry-After header field can be used with a 500 (Server Internal Error) or 503 (Service Unavailable) response to indicate how long the	section 20.33 page 177	181	we can write one test for UAC	UAC	As per rfc3261 the Retry-After header field can be used with a 500 (Server Internal Error) or 503 (Service	UAC MUST retry after Time mention in Retry -After Header field.	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>

	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 comment can be used to indicate additional information about the time of callback. 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)					Unavailable) 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. Here UAC sends INVITE to UAS. We as UAS sends 4xx response and a Retry-After header field containing Time(in sec) after which UAC SHOULD retry. UAC MUST retry after Time mention in Retry -After Header field.		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/2.0 486 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 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: 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

								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 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
119	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	section 21.4.23 page 189	182	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 responce.	UAC	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 responce.	UAC use one of the URI in Contact Header Field in responce.	F1: UAC->UAS INVITE sip:carl@example.comSIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:carl@example.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 F2: UAS->UAC SIP/2.0 485 Ambiguous Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:bob@biloxi.com>;tag=a6c85cf

	ambiguous Request-URIs. Example response to a request with the Request-URI sip:lee@example.com: SIP/2.0 485 Ambiguous 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> 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.							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 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 : UAC->UAS INVITE sip:carol.lee@example.com SIP/2.0 Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:carol.lee@example.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 F5: UAS->UAC SIP/2.0 200 OK Via:SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8 To: Bob <sip:carol.lee@example.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:carol.lee@example.com>;tag=a6c85cf From: Alice <sip:alice@atlanta.com>;tag=1928301774 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:

S.no RFC3261 Section Source Reason

1	A Request-Line contains a method name, a Request-URI, and the protocol version separated by a single space (SP) character.	section 7 page 28	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 + " " + self.rUri.create() + " " + self.protocol + "/" + self.version + "\r\n")
2	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.	section 13.2.2 page 81	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.
3	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	Same as above
4	If a fatal transport error is reported by the transport layer (generally, due to fatal ICMP errors in UDP or connection failures in TCP), the condition MUST be treated as a 503 (Service Unavailable) status code.	section 8.1.3.1 page 42	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.