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Technical Specification

**Technical Committee for IMS Network Testing (INT);
Network Integration Testing;
Part 3: Test Suite Structure and
Test Purposes (TSS&TP) for SIP-SIP**

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Foreword

This Technical Specification (TS) has been produced by IMS Network Testing (INT).

The present document is part 3 of a multi-part deliverable covering Network Integration Testing, as identified below:

Part 1: "Test Suite Structure and Test Purposes (TSS&TP) for SIP-ISDN";

Part 2: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) proforma specification";

Part 3: "Test Suite Structure and Test Purposes (TSS&TP) specification for SIP-SIP";

Part 4: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) for SIP-SIP".

1 Scope

The present document specifies the Test Suite Structure and Test Purposes (TSS&TP) for Network Integration Testing (NIT) to verify the overall compatibility of IMS networks. For IMS, SIP and SDP specific terminology, reference shall be made to ES 283 003 [1] and RFC 3261 [3] respectively".

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific.

- For a specific reference, subsequent revisions do not apply.
- Non-specific reference may be made only to a complete document or a part thereof and only in the following cases:
    - if it is accepted that it will be possible to use all future changes of the referenced document for the purposes of the referring document;
    - for informative references.

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NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long term validity.

2.1 Normative references

The following referenced documents are indispensable for the application of the present document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

[1] ETSI ES 283 003 (V2.6.1): "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Stage 3 [3GPP TS 24.229 [Release 7], modified]".

[2] ETSI TS 124 503 (V8.5.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; TISPAN; IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Stage 3 [3GPP TS 24.229 (Release 7), modified] (3GPP TS 24.503 version 8.5.0 Release 8)".

[3] IETF RFC 3261 (2002): "SIP: Session Initiation Protocol".

[4] ISO/IEC 9646-1 (1994): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 1: General concepts".

[5] ISO/IEC 9646-2 (1994): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 2: Abstract Test Suite specification".

[6] ISO/IEC 9646-3 (1998): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 3: The Tree and Tabular Combined Notation (TTCN)".

[7] Void.

[8] ISO/IEC 9646-5 (1994): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 5: Requirements on test laboratories and clients for the conformance assessment process".

[9] ISO/IEC 9646-7 (1995): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 7: Implementation Conformance Statements".

[10] ETSI TS 124 229 (V7.15.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229 version 7.15.0 Release 7)".

[11] Void.

[12] ETSI TS 124 504 (V8.5.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; TISPAN; PSTN/ISDN simulation services: Communication Diversion (CDIV); Protocol specification (3GPP TS 24.504 version 8.5.0 Release 8)".

[13] Void.

[14] ETSI TS 124 407 (V7.0.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); TISPAN; PSTN/ISDN simulation services; Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR); Protocol specification (3GPP TS 24.407 version 7.0.0 Release 7)".

[15] ETSI TS 124 410 (V7.0.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); TISPAN; NGN Signalling Control Protocol; Communication HOLD (HOLD) PSTN/ISDN simulation services; Protocol specification (3GPP TS 24.410 version 7.0.0 Release 7)".

[16] IETF RFC 2327 (1998): "SDP: Session Description Protocol".

[17] IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".

[18] IETF RFC 3311 (2002): "The Session Initiation Protocol UPDATE Method".

[19] ETSI TS 124 147 (V8.2.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3 (3GPP TS 24.147 version 8.2.0 Release 8)".

[20] Void.

[21] ETSI TS 124 615 (V8.2.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification (3GPP TS 24.615 version 8.2.0 Release 8)".

[22] ETSI TS 124 642 (V8.2.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Completion of Communications to Busy Subscriber (CCBS) and Completion of Communications by No Reply (CCNR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification (3GPP TS 24.642 version 8.2.0 Release 8)".

[23] ETSI TS 124 529 (V8.1.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); TISPAN; PSTN/ISDN simulation services: Explicit Communication Transfer (ECT); Protocol specification (3GPP TS 24.529 version 8.1.0 Release 8)".

[24] ETSI TS 124 508 (V8.1.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); PSTN/ISDN simulation services Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR); Protocol specification (3GPP TS 24.508 version 8.1.0 Release 8)".

[25] IETF RFC 5366: "Conference Establishment Using Request-Contained Lists in the Session Initiation Protocol (SIP)".

2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area**.** For non-specific references, the latest version of the referenced document (including any amendments) applies.

Not applicable.

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

For SIP and SDP specific terminology, reference shall be made to RFC 3261 [3] and RFC 2327 [16] respectively.

SIP precondition: Indicates the support of the SIP "precondition procedure" as defined in RFC 3312 [17].

test purpose: non-formal test description, mainly using text

NOTE: TSIs test description can be used as the basis for a formal test specification (e.g. Abstract Test Suite in TTCN). See ISO 9646 (all parts) [4] to [9].

The test purposes have been defined from the user's viewpoint and the abbreviation "UE" is used in the description. However, the detailed comments section uses the abbreviation "UA" for test system instances of the users.

3.2 Conventions for representation of SIP/SDP information

1) All letters of SIP method names are capitalised.

EXAMPLE 1: INVITE, INFO.

2) SIP header fields are identified by the unabbreviated header field name as defined in the relevant RFC, including capitalization and enclosed hyphens but excluding the following colon.

EXAMPLE 2: To, From, Call-ID.

3) Where it is necessary to refer with finer granularity to components of a SIP message, the component concerned is identified by the ABNF rule name used to designate it in the defining RFC (generally 25/RFC 3261 [3]), in plain text without surrounding angle brackets.

EXAMPLE 3: Request-URI, the userinfo portion of a sip: URI.

4) URI types are represented by the lower-case type identifier followed by a colon and the abbreviation "URI".

EXAMPLE 4: sip: URI, tel: URI.

5) SIP provisional responses and final responses other than 2XX are represented by the status code followed by the normal reason phrase for that status code, with initial letters capitalized.

EXAMPLE 5: 100 Trying, 484 Address Incomplete.

6) Because of potential ambiguity within a call flow about which request a 200 OK final response answers, 200 OK is always followed by the method name of the request.

EXAMPLE 6: 200 OK INVITE, 200 OK PRACK.

7) A particular line of an SDP session description is identified by the two initial characters of the line -- that is, the line type character followed by "=".

EXAMPLE 7: m=line, a=line.

8) Where it is necessary to refer with finer granularity to components of a session description, the component concerned is identified by its rule name in the ABNF description of the SDP line concerned, delimited with angle brackets.

EXAMPLE 8: The <media> and <fmt> components of the m= line.

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ABNF	Augmented Backus-Naur Form
ATS	Abstract Test Suite
CCBS	Completion of Communications to Busy Subscriber
CCNR	Completion of Communications by No Reply
CD	Communication Deflection
CDIV	Communication DIVersion
CDIVN	Communication DIVersion Notification
CFB	Communication Forwarding Busy
CFNL	Communication Forwarding on Not Logged-in
CFNR	Communication Forwarding No Replay
CFNRc	Communication Forwarding on subscriber Not Reachable
CFU	Communication Forwarding Unconditional
CONF	CONFerence
CW	Call Waiting
ECT	Explicit Communication Transfer
HOLD	communication HOLD
IUT	Implementation Under Test
NDUB	Network Determined User Busy
OIP	Originating Identification Presentation
OIR	Originating Identification Restriction
PIXIT	Protocol Implementation eXtra Information for Testing
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SUT	System Under Test
TIP	Terminating Identification Presentation
TIR	Terminating Identification Restriction
TP	Test Purpose
TSI	Test System Interface
TSS	Test Suite Structure
TTCN	Test and Test Control Notation
UA	User Agent
UDUB	User Determined User Busy
UE	User Equipment

4 Test Suite Structure (TSS)

4.1 SIP-SIP

C - Plane / U - Plane

Basic_Call			
	Successful		
		Normal call establishment	SS___XX___xx
		Codec negotiation	SS___CN___xx
		UPDATE	SS___XX__UP__xx
	Unsuccessful		SS___XX__Uxx
Supplementary_Services			
	OIP		SS___XXSS_OIPxx
	OIR		SS___XXSS_OIRxx
	TIP		SS___XXSS_TIPxx
	TIR		SS___XXSS_TIRxx
	HOLD		SS___XXSS_CHxx
	CDIV		
		CFU	SS___XXSS_CFUxx
		CFB	SS___XXSS_CFBxx
		CFNR	SS___XXSS_CFNRxx
		CFNRc	SS___XXSS_CFNRcxx
		CFNL	SS___XXSS_CFNLxx
		CD	SS___XXSS_CDxx
	CONF		
		CONF_CRE	SS___XXSS_CONF_CRExx
		CONF_IN	SS___XXSS_CONF_INVxx
		CONF_LEAV	SS___XXSS_CONF_LEAVxx
		CONF_REMOV	SS___XXSS_CONF_REMOVxx
	CW		SS___XXSS_CWxx
	CCBS		SS___XXSS_CCBSxx
	CCNR		SS___XXSS_CCNRxx
	ECT		SS___XXSS_ECT

5 Numbering Scheme

5.1 General description

Pos. 1: Network of the A-Subscriber

Pos. 2: Network of the B-Subscriber

Pos. 3: Network of the C-Subscriber

Pos. 4: Network of the D-Subscriber

Pos. 5: Network of the E-Subscriber

The following Network Codes apply:

_: No such network used (used e.g. for C-Subscriber in successful A to B Calls)

(underscore makes it easier to read the name)

P: PSTN

I: ISDN

S: SIP

(Extensions will be added when needed)

Pos. 6 and 7: Bearer- or Teleservice involved

XX: Defined per PIXIT value

NOTE: TSIs may be appropriate for Test Purposes (provided the Test Purpose states for which Bearer- and/or Tele Services it should be tested). It is however NOT appropriate for Test Cases since it would be detrimental to Test Automation.

SP: Speech

AU: 3,1 kHz Audio

UD: UDI

UT: UDI/TA

CN: Codec negotiation

DT: DTMF

UP: UPDATE Method

Pos. 8 and 9:

__: No Supplementary Services Involved / Successful

_U: No Supplementary Services Involved / Unsuccessful

SS: Supplementary Services Involved

5.2 Basic Call

Speech	IS___XX__XX

1	2	3	4	5	6	7	8	9	10	11
I	S	_	_	_	S	P	_	_	x	x

5.3 Supplementary Services

CLIP	IS___XXSSCLIP XX

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
I	S	_	_	_	X	X	S	S	C	L	I	P	x	X

6 Test purposes

The registration and application usage procedures in the ATS shall be compliant to RFC 3261 [3] and ES 283 003 [1] (modified TS 124 229 [10] and TS 124 503 [2]). The validation of the registration procedure is out of scope of the present document and will be part of the preambles used in the abstract test cases.

6.1 Test purposes for Basic Call

6.1.1 Test purposes for SIP-SIP, Basic call, Successful

6.1.1.1 Normal call establishment

SS___XX__01	**NGN reference to: RFC 3261 [3] TS 124 229 [10], clauses 5.1.3, 5.1.4**	
TSS reference:	SIP-SIP/Basic_call/Successful.	
Selection criteria:		
Test purpose:	Ensure that call establishment between UE A and UE B is handled correctly when reliable provisional responses and the precondition framework are **not** used. Ensure that the handling and mapping of the SDP parameters of the INVITE message is performed correctly. The call is released by the **called** user. Ensure that in the active call state the voice/data transfer on the media channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	INVITE: Require header **without** 100rel and precondition option tags sdp: PIXIT (Value should be taken from tables 1 and 2) 180 Ringing: Require header **without** 100rel **SIP UA A** **SUT** **SIP UA B** INVITE → → INVITE 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK Check media BYE ← ← BYE 200 OK BYE → → 200 OK BYE	

SS___XX__02	**NGN reference to: RFC 3261 [3], RFC 3312 [17] TS 124 229 [10], clauses 5.1.3, 5.1.4**	
TSS reference:	SIP-SIP/Basic_call/Successful.	
Selection criteria:		
Test purpose:	Ensure that call establishment between UE A and UE B is handled correctly when reliable provisional responses and the precondition framework are used. Ensure that the messages for the resource negotiation and reservation are delivered correctly. The call is released by the **called** user. Ensure that in the active call state the voice/data transfer on the media channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	INVITE: Supported header **with** 100rel and precondition option tags sdp: PIXIT (Value should be taken from tables 1 and 2) a=curr and a=des lines present 183 Session Progress: Require header **with** 100rel sdp: a=curr and a=des lines present UPDATE1 sdp: a=curr and a=des lines present	

Comments:	**SIP UA A**	**SUT**		**SIP UA B**
	INVITE	→	→	INVITE
	Start resource negotiation/reservation			
	183 Session Progress	←	←	183 Session Progress
	PRACK	→	→	PRACK
	200 OK PRACK	←	←	200 OK PRACK
	UPDATE	→	→	UPDATE1
	200 OK UPDATE1	←	←	200 OK UPDATE
	End resource negotiation/reservation			
	180 Ringing	←	←	180 Ringing
	PRACK	→	→	PRACK
	200 OK PRACK	←	←	200 OK PRACK
	200 OK INVITE	←	←	200 OK INVITE
	ACK	→	→	ACK
	Check media			
	BYE	←	←	BYE
	200 OK BYE	→	→	200 OK BYE

SS___XX__03	**NGN reference to:** **RFC 3261 [3]** **TS 124 229 [10], clauses 5.1.3, 5.1.4**	
TSS reference:	SIP-SIP/Basic_call/Successful.	
Selection criteria:		
Test purpose:	Ensure that call establishment between UE A and UE B is handled correctly when reliable provisional responses and the precondition framework are **not** used. Ensure that the handling and mapping of the SDP parameters of the INVITE message is performed correctly. The call is released by the **calling** user. Ensure that in the active call state the voice/data transfer on the media channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	INVITE: Require header **without** 100rel and precondition option tags sdp: PIXIT (Value should be taken from tables 1 and 2) 180 Ringing: Require header **without** 100rel	

	SIP UA A	**SUT**		**SIP UA B**
	INVITE	→	→	INVITE
	180 Ringing	←	←	180 Ringing
	200 OK INVITE	←	←	200 OK INVITE
	ACK	→	→	ACK
	Check media			
	BYE	→	→	BYE
	200 OK BYE	←	←	200 OK BYE

SS___XX__04	**NGN reference to:** **RFC 3261 [3], RFC 3312 [17]** **TS 124 229 [10], clauses 5.1.3, 5.1.4**	
TSS reference:	SIP-SIP/Basic_call/Successful.	
Selection criteria:		
Test purpose:	Ensure that call establishment between UE A and UE B is handled correctly when reliable provisional responses and the precondition framework are used. Ensure that the messages for the resource negotiation and reservation are delivered correctly. The call is released by the **calling** user. Ensure that in the active call state the voice/data transfer on the media channels is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	INVITE: Supported header **with** 100rel and precondition option tags sdp: PIXIT (Value should be taken from tables 1 and 2) a=curr and a=des lines present 183 Session Progress: Require header **with** 100rel sdp: a=curr and a=des lines present UPDATE1 sdp: a=curr and a=des lines present	
Comments:	**SIP UA A** **SUT** **SIP UA B** INVITE → → INVITE Start resource negotiation/reservation 183 Session Progress ← ← 183 Session Progress PRACK → → PRACK 200 OK PRACK ← ← 200 OK PRACK UPDATE → → UPDATE1 200 OK UPDATE1 ← ← 200 OK UPDATE End resource negotiation/reservation 180 Ringing ← ← 180 Ringing PRACK → → PRACK 200 OK PRACK ← ← 200 OK PRACK 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK Check media BYE → → BYE 200 OK BYE ← ← 200 OK BYE	

Table 1: Values for the test purpose SS___XX__01 to SS___XX__04

m= line				b= line	a= line
VA	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>
				See note	
VA_01	Audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A
VA_02	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000
VA_03	Audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A
VA_04	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000
VA_05	Image	Udptl	t38	N/A or up to 64 kbit/s	Based on T.38
VA_06	Image	Tcptl	t38	N/A or up to 64 kbit/s	Based on T.38
NOTE: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.					

Table 2: Values for test purposes SS___XX__01 and SS___XX__04

VARIABLE	PT	Encoding	media type	clock rate	channels
VA_01	0	PCMU	A	8,000	1
VA_02	3	GSM	A	8,000	1
VA_03	4	G723	A	8,000	1
VA_04	5	DVI4	A	8,000	1
VA_05	6	DVI4	A	16,000	1
VA_06	7	LPC	A	8,000	1
VA_07	8	PCMA	A	8,000	1
VA_08	9	G722	A	8,000	1
VA_09	10	L16	A	44,100	2
VA_10	11	L16	A	44,100	1
VA_13	12	QCELP	A	8,000	1
VA_12	13	CN	A	8,000	1
VA_13	14	MPA	A	90,000	
VA_14	15	G728	A	18,000	1
VA_15	16	DVI4	A	11,025	1
VA_16	17	DVI4	A	22,050	1
VA_17	18	G729	A	8,000	1
VA_18	Dyn	G726-40	A	8,000	1
VA_19	Dyn	G726-32	A	8,000	1
VA_20	Dyn	G726-24	A	8,000	1
VA_21	Dyn	G726-16	A	8,000	1
VA_22	Dyn	G729D	A	8,000	1
VA_23	Dyn	G729E	A	8,000	1
VA_24	Dyn	GSM-EFR	A	8,000	1
VA_25	25	CelB	V	90,000	
VA_26	26	JPEG	V	90,000	
VA_27	28	Nv	V	90,000	
VA_28	31	H261	V	90,000	
VA_29	32	MPV	V	90,000	
VA_30	33	MP2T	V	90,000	
VA_31	34	H263	V	90,000	
VA_32	Dyn	H263-1998	V	90,000	

6.1.1.2 Codec negotiation

<table>
<tr><td>SS___CN__01</td><td colspan="3">NGN reference to:
RFC 3261 [3]
TS 124 229 [10], clauses 5.1.3, 5.1.4</td><td colspan="2"></td></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-SIP/Basic_call/Codec negotiation</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the SUT, when the calling user decides during a session which was set-up without using the precondition mechanism to change the characteristics of the media session by sending a re-INVITE request,
transports the re-INVITE request and the related 200 OK and ACK messages correctly.
Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">re-INVITE:
sdp: PIXIT (Value should be taken from tables 1 and 2)</td></tr>
<tr><td rowspan="12">Comments:</td><td>SIP UA A</td><td></td><td>SUT</td><td></td><td>SIP UA B</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>re-INVITE</td><td>→</td><td></td><td>→</td><td>re-INVITE</td></tr>
<tr><td>200 OK re-INVITE</td><td>←</td><td></td><td>←</td><td>200 OK re-INVITE</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Check media</td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
</table>

SS___CN__02	**NGN reference to: RFC 3261 [3], RFC 3312 [17] TS 124 229 [10], clauses 5.1.3, 5.1.4**		
TSS reference:	SIP-SIP/Basic_call/Codec negotiation		
Selection criteria:			
Test purpose:	Ensure that the IUT, when the **calling** user decides during a session which was set-up **with** using the precondition mechanism to change the characteristics of the media session by sending a re-INVITE request, transports the re-INVITE request and the related 200 OK and ACK messages correctly. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).		
SIP Parameter values:	re-INVITE: sdp: PIXIT (Value should be taken from tables 1 and 2)		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→ →	INVITE
		Start resource negotiation/reservation	
		←	183 Session Progress SDP
	183 Session Progress SDP	←	
	PRACK	→ →	PRACK
	200 OK PRACK	← ←	200 OK PRACK
	UPDATE	→ →	UPDATE
	200 OK UPDATE	← ←	200 OK UPDATE
		End resource negotiation/reservation	
	180 Ringing	← ←	180 Ringing
	PRACK	→ →	PRACK
	200 OK PRACK	← ←	200 OK PRACK
	200 OK INVITE	← ←	200 OK INVITE
	ACK	→ →	ACK
	re-INVITE	→ →	re-INVITE
	200 OK re-INVITE	← ←	200 OK re-INVITE
	ACK	→ →	ACK
		Check media	
	BYE	← ←	BYE
	200 OK BYE	→ →	200 OK BYE
NOTE: Re-Invite may need precondition, too (but is out of scope of this test case).			

SS___CN__03	**NGN reference to: RFC 3261 [3] TS 124 229 [10], clauses 5.1.3, 5.1.4**		
TSS reference:	SIP-SIP/Basic_call/Codec negotiation		
Selection criteria:			
Test purpose:	Ensure that the SUT, when the **called** user decides during a session which was set-up **without** using the precondition mechanism to change the characteristics of the media session by sending a re-INVITE, transports the re-INVITE request and the related 200 OK and ACK messages correctly. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).		
SIP Parameter values:	re-INVITE: sdp: PIXIT (Value should be taken from tables 1 and 2)		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→ →	INVITE
	180 Ringing	← ←	180 Ringing
	200 OK INVITE	← ←	200 OK INVITE
	ACK	→ →	ACK
	re-INVITE	← ←	re-INVITE
	200 OK re-INVITE	→ →	200 OK re-INVITE
	ACK	← ←	ACK
		Check media	
	BYE	← ←	BYE
	200 OK BYE	→ →	200 OK BYE

SS___CN__04	**NGN reference to:** **RFC 3261 [3]** **TS 124 229 [10], clauses 5.1.3, 5.1.4**				
TSS reference:	SIP-SIP/Basic_call/Codec negotiation				
Selection criteria:					
Test purpose:	Ensure that the IUT, when the **called** user decides during a session which was set-up **with** using the precondition mechanism to change the characteristics of the media session by sending a re-INVITE, transports the re-INVITE request and the related 200 OK and ACK messages correctly. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).				
SIP Parameter values:	re-INVITE: sdp: PIXIT (Value should be taken from tables 1 and 2)				
Comments:	**SIP UA A**		**SUT**		**SIP UA B**
	INVITE	→		→	INVITE
			Start resource negotiation/reservation		
				←	183 Session Progress SDP
	183 Session Progress SDP	←			
	PRACK	→		→	PRACK
	200 OK PRACK	←		←	200 OK PRACK
	UPDATE	→		→	UPDATE
	200 OK UPDATE	←		←	200 OK UPDATE
			End resource negotiation/reservation		
	180 Ringing	←		←	180 Ringing
	PRACK	→		→	PRACK
	200 OK PRACK	←		←	200 OK PRACK
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
	re-INVITE re-INVITE	←		←	re-INVITE re-INVITE
	200 OK	→		→	200 OK
	ACK	←		←	ACK
			Check media		
	BYE	←		←	BYE
	200 OK BYE	→		→	200 OK BYE
NOTE: Re-Invite may need precondition, too (but is out of scope of this test case).					

SS___CN__05	**NGN reference to:** **RFC 3261 [3]** **TS 124 229 [10], clauses 5.1.3, 5.1.4**				
TSS reference:	SIP-SIP/Basic_call/Codec negotiation				
Selection criteria:					
Test purpose:	Ensure that the SUT can correctly transport an SDP answer related to the SDP offer in the INVITE request in the 180 Ringing message, which is sent reliably. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).				
SIP Parameter values:	INVITE: sdp: PIXIT (Value should be taken from tables 1 and 2) Supported header **with** 100rel option tag 180 Ringing: sdp: PIXIT (Value should be taken from tables 1 and 2) Require header **with** 100rel option tag				
Comments:	**SIP UA A**		**SUT**		**SIP UA B**
	INVITE	→		→	INVITE
	180 Ringing with SDP answer	←		←	180 Ringing with SDP answer
	PRACK	→		→	PRACK
	200 OK PRACK	←		←	200 OK PRACK
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
			Check media		
	BYE	←		←	BYE
	200 OK BYE	→		→	200 OK BYE

SS___CN__06	**NGN reference to: RFC 3261 [3] TS 124 229 [10], clauses 5.1.3, 5.1.4**		
TSS reference:	SIP-SIP/Basic_call/Codec negotiation		
Selection criteria:			
Test purpose:	Ensure that the SUT can correctly transport an SDP answer related to the SDP offer in the INVITE request in the 183 Session Progress message, which is sent reliably. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).		
SIP Parameter values:	INVITE: sdp: PIXIT (Value should be taken from tables 1 and 2) Supported header **with** 100rel option tag 183 Session Progress: sdp: PIXIT (Value should be taken from tables 1 and 2) Require header **with** 100rel option tag		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→ →	INVITE
	183 Session Progress with SDP answer	← ←	183 Session Progress with SDP answer
	PRACK	→ →	PRACK
	200 OK PRACK	← ←	200 OK PRACK
	180 Ringing	← ←	180 Ringing
	200 OK INVITE	← ←	200 OK INVITE
	ACK	→ →	ACK
		Check media	
	BYE	← ←	BYE
	200 OK BYE	→ →	200 OK BYE

SS___CN__07	**NGN reference to: RFC 3261 [3] TS 124 229 [10], clauses 5.1.3, 5.1.4**		
TSS reference:	SIP-SIP/Basic_call/Codec negotiation		
Selection criteria:			
Test purpose:	Ensure that the SUT can correctly transport an SDP answer related to the SDP offer in the INVITE request in the 200 OK message. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).		
SIP Parameter values:	INVITE: sdp: PIXIT (Value should be taken from tables 1 and 2) Supported header **with** 100rel option tag 200 OK: sdp: PIXIT (Value should be taken from tables 1 and 2)		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→ →	INVITE
	180 Ringing	← ←	180 Ringing
	200 OK INVITE with SDP answer	← ←	200 OK INVITE with SDP answer
	ACK	→ →	ACK
		Check media	
	BYE	← ←	BYE
	200 OK BYE	→ →	200 OK BYE

6.1.1.3 UPDATE method

SS___UP__01	**NGN reference to: RFC 3261 [3], RFC 3311 [18] TS 124 229 [10], clauses 5.1.3, 5.1.4**				
TSS reference:	SIP-SIP/Basic_call/update				
Selection criteria:					
Test purpose:	Ensure that the SUT, when the **calling** user decides during a session which was set-up **without** using the precondition mechanism to change the characteristics of the media session by sending an UPDATE request, transports the UPDATE request and the related 200 OK and ACK messages correctly. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).				
SIP Parameter values:	UPDATE: sdp: PIXIT (Value should be taken from tables 1 and 2)				
Comments:	**SIP UA A**		**SUT**		**SIP UA B**
	INVITE	→		→	INVITE
	180 Ringing	←		←	180 Ringing
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
	UPDATE	→		→	UPDATE
	200 OK UPDATE	←		←	200 OK UPDATE
			Check media		
	BYE	←		←	BYE
	200 OK BYE	→		→	200 OK BYE

SS___UP__02	**NGN reference to: RFC 3261 [3], RFC 3312 [17], RFC 3311 [18] TS 124 229 [10], clauses 5.1.3, 5.1.4**				
TSS reference:	SIP-SIP/Basic_call/update				
Selection criteria:					
Test purpose:	Ensure that the IUT, when the **calling** user decides during a session which was set-up **with** using the precondition mechanism to change the characteristics of the media session by sending an UPDATE request, transports the UPDATE request and the related 200 OK message correctly. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).				
SIP Parameter values:	UPDATE: sdp: PIXIT (Value should be taken from tables 1 and 2)				
Comments:	**SIP UA A**		**SUT**		**SIP UA B**
	INVITE	→		→	INVITE
			Start resource negotiation/reservation		
				←	183 Session Progress SDP
	183 Session Progress SDP	←			
	PRACK	→		→	PRACK
	200 OK PRACK	←		←	200 OK PRACK
	UPDATE	→		→	UPDATE
	200 OK UPDATE	←		←	200 OK UPDATE
			End resource negotiation/reservation		
	180 Ringing	←		←	180 Ringing
	PRACK	→		→	PRACK
	200 OK PRACK	←		←	200 OK PRACK
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
	UPDATE	→		→	UPDATE
	200 OK UPDATE	←		←	200 OK UPDATE
			Check media		
	BYE	←		←	BYE
	200 OK BYE	→		→	200 OK BYE
NOTE: UPDATE after session establishment may need precondition, too (but is out of scope of this test case).					

SS___UP__03	**NGN reference to: RFC 3261 [3], RFC 3311 [18] TS 124 229 [10], clauses 5.1.3, 5.1.4**				
TSS reference:	SIP-SIP/Basic_call/update				
Selection criteria:					
Test purpose:	Ensure that the SUT, when the **called** user decides during a session which was set-up **without** using the precondition mechanism to change the characteristics of the media session by sending an UPDATE request, transports the UPDATE request and the related and ACK message correctly. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).				
SIP Parameter values:	UPDATE: sdp: PIXIT (Value should be taken from tables 1 and 2)				
Comments:	**SIP UA A**		**SUT**		**SIP UA B**
	INVITE	→		→	INVITE
	180 Ringing	←		←	180 Ringing
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
	UPDATE	←		←	UPDATE
	200 OK UPDATE	→		→	200 OK UPDATE
			Check media		
	BYE	←		←	BYE
	200 OK BYE	→		→	200 OK BYE

SS___UP__04	**NGN reference to: RFC 3261 [3], RFC 3312 [17], RFC 3311 [18] TS 124 229 [10], clauses 5.1.3, 5.1.4**				
TSS reference:	SIP-SIP/Basic_call/update				
Selection criteria:					
Test purpose:	Ensure that the IUT, when the **called** user decides during a session which was set-up **with** using the precondition mechanism to change the characteristics of the media session by sending an UPDATE request, transports the UPDATE request and the related 200 OK message correctly. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).				
SIP Parameter values:	UPDATE: sdp: PIXIT (Value should be taken from tables 1 and 2)				
Comments:	**SIP UA A**		**SUT**		**SIP UA B**
	INVITE	→		→	INVITE
			Start resource negotiation/reservation		
				←	183 Session Progress SDP
	183 Session Progress SDP	←			
	PRACK	→		→	PRACK
	200 OK PRACK	←		←	200 OK PRACK
	UPDATE	→		→	UPDATE
	200 OK UPDATE	←		←	200 OK UPDATE
			End resource negotiation/reservation		
	180 Ringing	←		←	180 Ringing
	PRACK	→		→	PRACK
	200 OK PRACK	←		←	200 OK PRACK
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
	UPDATE	←		←	UPDATE
	200 OK UPDATE	→		→	200 OK UPDATE
			Check media		
	BYE	←		←	BYE
	200 OK BYE	→		→	200 OK BYE
NOTE: UPDATE after session establishment may need precondition, too (but is out of scope of this test case).					

<table>
<tr><td>SS___UP__05</td><td colspan="3">NGN reference to:
RFC 3261 [3], RFC 3312 [17],
RFC 3311 [18]
TS 124 229 [10], clauses 5.1.3, 5.1.4</td><td colspan="2"></td></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-SIP/Basic_call/update</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the IUT, after an SDP offer in an INVITE request from the calling user has been answered in a reliably sent 180 Ringing message by the called user, when the calling user decides before the end of session establishment to change the characteristics of the media session by sending an UPDATE request, transports the UPDATE request and the related 200 OK message correctly.
Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">INVITE:
Allow including UPDATE
Supported header include 100rel
sdp offer1

180 Ringing:
Allow including UPDATE
Require header include 100rel
sdp answer1

UPDATE:
sdp offer2

200 OK UPDATE:
sdp answer2</td></tr>
<tr><td rowspan="2">Comments:</td><td>SIP UA A</td><td></td><td>SUT</td><td></td><td>SIP UA B</td></tr>
<tr><td>INVITE (sdp offer1)
180 Ringing
(sdp answer1)
PRACK
200 OK PRACK
UPDATE (sdp offer2)
200 OK UPDATE
(sdp answer2)
200 OK INVITE
ACK

BYE
200 OK BYE</td><td>→
←

→
←
→
←

←
→

←
→</td><td>

Check media</td><td>→
←

→
←
→
←

←
→

←
→</td><td>INVITE (sdp offer1)
180 Ringing
(sdp answer1)
PRACK
200 OK PRACK
UPDATE (sdp offer2)
200 OK UPDATE
(sdp answer2)
200 OK INVITE
ACK

BYE
200 OK BYE</td></tr>
</table>

SS___UP__06	**NGN reference to:** **RFC 3261 [3], RFC 3312 [17],** **RFC 3311 [18]** **TS 124 229 [10], clauses 5.1.3, 5.1.4**	
TSS reference:	SIP-SIP/Basic_call/update	
Selection criteria:		
Test purpose:	Ensure that the IUT, after an SDP offer in an INVITE request from the calling user has been answered in a reliably sent 180 Ringing message by the called user, when the **called** user decides before the end of session establishment to change the characteristics of the media session by sending an UPDATE request, transports the UPDATE request and the related 200 OK message correctly. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	INVITE: Allow including UPDATE Supported header include 100rel sdp offer1 180 Ringing: Allow including UPDATE Require header include 100rel sdp answer1 UPDATE: sdp offer2 200 OK UPDATE: sdp answer2	

Comments:	**SIP UA A**		**SUT**		**SIP UA B**
	INVITE (sdp offer1)	→		→	INVITE (sdp offer1)
	180 Ringing (sdp answer1)	←		←	180 Ringing (sdp answer1)
	PRACK	→		→	PRACK
	200 OK PRACK	←		←	200 OK PRACK
	UPDATE (sdp offer2)	←		←	UPDATE (sdp offer2)
	200 OK UPDATE (sdp answer2)	→		→	200 OK UPDATE (sdp answer2)
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
			Check media		
	BYE	←		←	BYE
	200 OK BYE	→		→	200 OK BYE

<table>
<tr><td>SS___UP__07</td><td colspan="2">NGN reference to:
RFC 3261 [3], RFC 3312 [17],
RFC 3311 [18]
TS 124 229 [10], clauses 5.1.3, 5.1.4</td><td colspan="2"></td></tr>
<tr><td>TSS reference:</td><td colspan="4">SIP-SIP/Basic_call/update</td></tr>
<tr><td>Selection criteria:</td><td colspan="4"></td></tr>
<tr><td>Test purpose:</td><td colspan="4">Ensure that the IUT, after an INVITE request without SDP offer has been sent by the calling user and an SDP offer from the called user reliably sent in a 180 Ringing message has been answered by the calling user in the PRACK message, when the calling user decides before the end of session establishment to change the characteristics of the media session by sending an UPDATE request, transports the UPDATE request and the related 200 OK message correctly.
Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="4">INVITE:
Allow including UPDATE
Supported header include 100rel
sdp not present1

180 Ringing:
Allow including UPDATE
Require header include 100rel
sdp offer1

PRACK:
sdp answer1

UPDATE:
sdp offer2

200 OK UPDATE:
sdp answer2</td></tr>
<tr><td rowspan="2">Comments:</td><td>SIP UA A</td><td></td><td>SUT</td><td>SIP UA B</td></tr>
<tr><td>INVITE (no sdp)
180 Ringing (sdp offer1)
PRACK(sdp answer1)
200 OK PRACK
UPDATE (sdp offer2)
200 OK UPDATE
(sdp answer2)
200 OK INVITE
ACK

BYE
200 OK BYE</td><td>→
←
→
←
→
←

←
→

←
→</td><td>

Check media</td><td>→ INVITE (no sdp)
← 180 Ringing (sdp offer1)
→ PRACK(sdp answer1)
← 200 OK PRACK
→ UPDATE (sdp offer2)
← 200 OK UPDATE
(sdp answer2)
← 200 OK INVITE
→ ACK

← BYE
→ 200 OK BYE</td></tr>
</table>

SS___UP__08	**NGN reference to: RFC 3261 [3], RFC 3312 [17], RFC 3311 [18] TS 124 229 [10], clauses 5.1.3, 5.1.4**			
TSS reference:	SIP-SIP/Basic_call/update			
Selection criteria:				
Test purpose:	Ensure that the IUT, after an INVITE request without SDP offer has been sent by the calling user and an SDP offer from the called user reliably sent in a 180 Ringing message has been answered by the calling user in the PRACK message, when the **called** user decides before the end of session establishment to change the characteristics of the media session by sending an UPDATE request, transports the UPDATE request and the related 200 OK message correctly. Ensure that the voice/data transfer on the media channels with the re-negotiated media is performed correctly (e.g. testing QoS parameters).			
SIP Parameter values:	INVITE: Allow including UPDATE Supported header:100rel sdp not present1 180 Ringing: Allow including UPDATE Require header include 100rel sdp offer1 PRACK: sdp answer1 UPDATE: sdp offer2 200 OK UPDATE: sdp answer2			
Comments:	**SIP UA A**		**SUT**	**SIP UA B**
	INVITE (no sdp)	➔	➔	INVITE (no sdp)
	180 Ringing (sdp offer1)	←	←	180 Ringing (sdp offer1)
	PRACK(sdp answer1)	➔	➔	PRACK(sdp answer1)
	200 OK PRACK	←	←	200 OK PRACK
	UPDATE (sdp offer2)	←	←	UPDATE (sdp offer2)
	200 OK UPDATE (sdp answer2)	➔	➔	200 OK UPDATE (sdp answer2)
	200 OK INVITE	←	←	200 OK INVITE
	ACK	➔	➔	ACK
			Check media	
	BYE	←	←	BYE
	200 OK BYE	➔	➔	200 OK BYE

6.1.2 Test purposes for SIP-SIP, Basic call, Unsuccessful

SS___XX_U01	**NGN reference to: RFC 3261 [3] TS 124 229 [10], clause 5.2.6.3**			
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.			
Selection criteria:				
Test purpose:	Ensure that the SUT delivers a 503 Service Unavailable message from the called to the calling user.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT			
Comments:	**SIP UA A**		**SUT**	**SIP UA B**
	INVITE	➔	➔	INVITE
	503 Service Unavailable	←	←	503 Service Unavailable
	ACK	➔	➔	ACK

SS___XX_U02	**NGN reference to: RFC 3261 [3] TS 124 229 [10], clause 5.2.6.3**			
TSS reference:	SIP-SIP/Basic_call/Unsuccessful			
Selection criteria:				
Test purpose:	Ensure that the SUT delivers a 486 Busy Here message from the called to the calling user.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT			
Comments:	**SIP UA A**		**SUT**	**SIP UA B**
	INVITE 486 Busy Here ACK	→ ← →	→ ← →	INVITE 486 Busy Here ACK

SS___XX_U03	**NGN reference to: RFC 3261 [3]**			
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.			
Selection criteria:				
Test purpose:	Ensure that when there is no answer from the called user (there is no response to the INVITE messages), the SUT initiates call clearing to the calling user with a 480 Temporarily Unavailable or 408 Request Timeout message.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT			
Comments:	**SIP UA A**		**SUT**	**SIP UA B**
	INVITE 100 Trying 480 Temporarily Unavailable or 408 Request Timeout ACK	→ ← ← →	→ → → → → → → →	INVITE INVITE INVITE INVITE INVITE INVITE INVITE INVITE
NOTE: No 100 Trying response by UA-B.				

SS___XX_U04	**NGN reference to: RFC 3261 [3] TS 124 229 [10], clause 5.2.6.3**			
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.			
Selection criteria:				
Test purpose:	Ensure that the SUT delivers a 480 Temporarily Unavailable message from the alerting called user to the calling user (do not disturb service).			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT			
Comments:	**SIP UA A**		**SUT**	**SIP UA B**
	INVITE 180 Ringing 480 Temporary unavaible ACK	→ ← ← →	→ ← ← →	INVITE 180 Ringing 480 Temporary unavaible ACK

SS___XX_U05	**NGN reference to: RFC 3261 [3] TS 124 229 [10]**		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.		
Selection criteria:			
Test purpose:	Ensure that when the calling user clears the call with a CANCEL message before receiving an answer to the previously sent INVITE request from the called user, the SUT delivers the CANCEL message to the called user.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→ →	INVITE
		→	INVITE (may be repeated)
	CANCEL	→ →	CANCEL
	200 OK CANCEL	← ←	200 OK CANCEL
	487 Request Terminated	← ←	487 Request Terminated
	ACK	→ →	ACK
NOTE: No 100 Trying response by UA-B.			

SS___XX_U06	**NGN reference to: RFC 3261 [3] TS 124 229 [10]**		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.		
Selection criteria:			
Test purpose:	Ensure that the IUT, when the **calling** user decides during a session to change the characteristics of the media session by sending a re-INVITE request and the Re-INVITE is rejected by the **called** user with a 488 Not Acceptable Here, delivers the 488 Not Acceptable Here to the **calling** user. Ensure that the voice/data transfer on the media channels with the original media is still performed correctly (e.g. testing QoS parameters).		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT		
Comments:	**SIP**	**SUP**	**SIP**
	INVITE	→ →	INVITE
	180 Ringing	← ←	180 Ringing
	200 OK INVITE	← ←	200 OK INVITE
	ACK	→ →	ACK
	Re-INVITE	→ →	Re-INVITE offer
	488 Not Acceptable Here	← ←	488 Not Acceptable Here
		Communication	
	BYE	← ←	BYE
	200 OK BYE		200 OK BYE

SS___XX_U07	**NGN reference to: RFC 3261 [3] TS 124 229 [10]**			
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.			
Selection criteria:				
Test purpose:	Ensure that the IUT, when the **called** user decides during a session to change the characteristics of the media session by sending a re-INVITE request and the Re-INVITE is rejected by the **calling** user with a 488 Not Acceptable Here, delivers the 488 Not Acceptable Here to the **called** user. Ensure that the voice/data transfer on the media channels with the original media is still performed correctly (e.g. testing QoS parameters).			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition.			
Comments:	**SIP**		**SUT**	**SIP**
	INVITE	→	→	INVITE
	180 Ringing	←	←	180 Ringing
	200 OK INVITE	←	←	200 OK INVITE
	ACK	→	→	ACK
	Re-INVITE	←	←	Re-INVITE
	488 Not Acceptable Here	→	→	488 Not Acceptable Here
			Communication	
	BYE	←	←	BYE
	200 OK BYE			200 OK BYE

SS___XX_U08	**NGN reference to: RFC 3261 [3] TS 124 229 [10]**			
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.			
Selection criteria:				
Test purpose:	Ensure that when there is no answer from the called user ("*no user responding*"), the SUT initiates call clearing to the called user with CANCEL and to the calling user with CANCEL.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT			
Comments:	**SIP**		**SUT**	**SIP**
	INVITE	→	→	INVITE
	180 Ringing	←	←	180 Ringing
			Timeout timer C	
	408/480	←	→	CANCEL
	ACK	→	←	200 OK CANCEL
			←	487 Request Terminated
			→	ACK

6.2 Test purposes for SIP-SIP, Supplementary services

6.2.1 Test purposes for OIP

SS___XXSS_OIP01	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIP		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "not restricted". The terminating user subscribes to OIP.		
Test purpose:	Ensure that, when no **P-Preferred-Identity** header field is provided by the originating UE in the INVITE request, the terminating user receives a **P-Asserted-Identity** based on the default public user identity associated with the originating UE.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →		→ INVITE

SS___XXSS_OIP02	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "not restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**none**" and no **P-Preferred-Identity** header field is provided by the originating UE, the terminating user receives a **P-Asserted-Identity** based on the default public user identity associated with the originating UE.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**none**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →		→ INVITE

SS___XXSS_OIP03	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**none**" and no **P-Preferred-Identity** header field is provided by the originating UE, the terminating user receives a **P-Asserted-Identity** based on the default public user identity associated with the originating UE.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**none**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →		→ INVITE

SS___XXSS_OIP04	OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.12		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "not restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when no Privacy header field is inserted and a **P-Preferred-Identity** header field is provided by the originating UE, but the identity information in the P-Preferred-Identity does not match with the set of registered public identities of the originating UE, the terminating user receives a **P-Asserted-Identity** based on the default public user identity associated with the originating UE.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →		→ INVITE

SS___XXSS_OIP05	OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.12		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "not restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**none**" and a **P-Preferred-Identity** header field is provided by the originating UE, but the identity information in the P-Preferred-Identity does not match with the set of registered public identities of the originating UE, the terminating user receives a **P-Asserted-Identity** based on the default public user identity associated with the originating UE.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**none**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →		→ INVITE

SS___XXSS_OIP06	OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.12		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**none**" and a **P-Preferred-Identity** header field is provided by the originating UE, but the identity information in the P-Preferred-Identity does not match with the set of registered public identities of the originating UE, the terminating user receives a **P-Asserted-Identity** based on the default public user identity associated with the originating UE.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**none**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →		→ INVITE

SS___XXSS_OIP07	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "not restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when no Privacy header field is inserted and a **P-Preferred-Identity** header field is provided by the originating UE (the identity information in the P-Preferred-Identity must be present in the set of registered public identities of the originating UE and it shall be different from the default public user identity), the terminating UE receives a **P-Asserted-Identity** based on the information provided by the originating UE.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→	→ INVITE

SS___XXSS_OIP08	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "not restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**none**" and a **P-Preferred-Identity** header field is provided by the originating UE (the identity information in the P-Preferred-Identity must be present in the set of registered public identities of the originating UE and it shall be different from the default public user identity), the terminating UE receives a **P-Asserted-Identity** based on the information provided by the originating UE.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**none**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→	→ INVITE

SS___XXSS_OIP09	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**none**" and a **P-Preferred-Identity** header field is provided by the originating UE (the identity information in the P-Preferred-Identity must be present in the set of registered public identities of the originating UE and it shall be different from the default public user identity), the terminating UE receives a **P-Asserted-Identity** based on the information provided by the originating UE.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**none**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→	→ INVITE

SS___XXSS_OIP10	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.		
Selection criteria:	The terminating user is not subscribed to OIP service.		
Test purpose:	Ensure that, for any INVITE request, the terminating user receives no **P-Asserted-Identity** header field and no Privacy header field.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→	→ INVITE

6.2.2 Test purposes for OIR

SS___XXSS_OIR01	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.1.2, 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "restricted". Also, the restricted type is set to "*restrict the asserted identity*" (see table 1, TS 124 407 [14], clause 4.3.1.2). The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when no Privacy header field is inserted by the originating UE in the INVITE request, the terminating UE receives an INVITE message where the From header field is set to "anonymous", the Privacy header field is set to "**id**" and no P-Asserted-Identity header is received.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →		→ INVITE

SS___XXSS_OIR02	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.1.2, 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "restricted". Also, the restricted type is set to "*restrict all private information appearing in headers*" (see table 1, TS 124 407 [14], clause 4.3.1.2). The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when no Privacy header field is inserted by the originating UE in the INVITE request, the terminating UE receives an INVITE message where the From header field is set to "anonymous", the Privacy header field is set to "**header**" and no P-Asserted-Identity header is received.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →		→ INVITE

SS___XXSS_OIR03	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**id**" by the originating UE in the INVITE request, the terminating UE receives an INVITE message where the From header field is set to "anonymous", the Privacy header field is set to "**id**" or "**header**" and no P-Asserted-Identity header is received.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**id**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →		→ INVITE

SS___XXSS_OIR04	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**header**" by the originating UE in the INVITE request, the terminating UE receives an INVITE message where the From header field is set to "anonymous", the Privacy header field is set to "**id**" or "**header**" and no P-Asserted-Identity header is received.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**header**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →	→ INVITE	

SS___XXSS_OIR05	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "not restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**id**" and the From header field is set to "anonymous" by the originating UE in the INVITE request, the terminating UE receives an INVITE message where the From header field is set to "anonymous", the Privacy header field is set to "**id**" and no P-Asserted-Identity header is received.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**id**" From header field is set to: *From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag= xxxxxxx*		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →	→ INVITE	

SS___XXSS_OIR06	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The originating user subscribes to OIR "temporary mode" default "not restricted". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**header**" and the From header field is set to "anonymous" by the originating UE in the INVITE request, the terminating UE receives an INVITE message where the From header field is set to "anonymous", the Privacy header field is set to "**header**" and no P-Asserted-Identity header is received.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**header**" From header field is set to: *From: "Anonymous" <sip:anonymous@anonymous.invalid>;tag= xxxxxxx*		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →	→ INVITE	

SS___XXSS_OIR07	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**	
TSS reference:	SIP-SIP/SupplementaryServices/OIR.	
Selection criteria:	The originating user subscribes to OIR permanent mode. Also, the restricted type is set to "*restrict the asserted identity*" (see table 1, TS 124 407 [14], clause 4.3.1.2). The terminating user subscribes to OIP service.	
Test purpose:	Ensure that, when no Privacy header field is inserted by the originating UE in the INVITE request, the terminating UE receives an INVITE message where the From header field is set to "anonymous", the Privacy header field is set to "**id**" and no P-Asserted-Identity header is received.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	
Comments:	**SIP UA A** **SUT** **SIP UA B**	
	INVITE ➔ ➔ INVITE	

SS___XXSS_OIR08	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**	
TSS reference:	SIP-SIP/SupplementaryServices/OIR.	
Selection criteria:	The originating user subscribes to OIR permanent mode. Also, the restricted type is set to "*restrict all private information appearing in headers*" (see table 1, TS 124 407 [14], clause 4.3.1.2). The terminating user subscribes to OIP service.	
Test purpose:	Ensure that, when no Privacy header field is inserted by the originating UE in the INVITE request, the terminating UE receives an INVITE message where the From header field is set to "anonymous", the Privacy header field is set to "**header**" and no P-Asserted-Identity header is received.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	
Comments:	**SIP UA A** **SUT** **SIP UA B**	
	INVITE ➔ ➔ INVITE	

SS___XXSS_OIR09	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**	
TSS reference:	SIP-SIP/SupplementaryServices/OIR.	
Selection criteria:	The originating user subscribes to OIR permanent mode. The terminating user subscribes to OIP service.	
Test purpose:	Ensure that, when the Privacy header field is set to "**id**" by the originating UE in the INVITE request, the terminating UE receives an INVITE message where the From header field is set to "anonymous", the Privacy header field is set to "**id**" or "**header**" and no P-Asserted-Identity header is received.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**id**"	
Comments:	**SIP UA A** **SUT** **SIP UA B**	
	INVITE ➔ ➔ INVITE	

SS___XXSS_OIR10	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The originating user subscribes to OIR permanent mode. The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**header**" by the originating UE in the INVITE request, the terminating UE receives an INVITE message where the From header field is set to "anonymous", the Privacy header field is set to "**id**" or "**header**" and no P-Asserted-Identity header is received.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**header**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →	→	INVITE

SS___XXSS_OIR11	**OIP/OIR reference to: TS 124 407 [14], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/OIP		
Selection criteria:	The originating user subscribes to OIR "permanent mode". The terminating user subscribes to OIP service.		
Test purpose:	Ensure that, when the Privacy header field is set to "**none**" by the originating UE in the INVITE request, the terminating UE receives an INVITE message where the From header field is set to "anonymous", the Privacy header field is set to "**id**" or "**header**" and no P-Asserted-Identity header is received.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**none**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →	→	INVITE

6.2.3 Test purposes for TIP

SS___XXSS_TIP01	**TIP/TIR reference to: TS 124 508 [24], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/TIP.		
Selection criteria:	The originating user subscribes to TIP service.		
Test purpose:	Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request: the originating UE receives, in the 2xx SIP response, a **P-Asserted-Identity** header field with a valid public user identity of the terminating UE.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE →	→	INVITE
	180 Ringing ←	←	180 Ringing
	200 OK INVITE (P-Asserted-Identity) ←	←	200 OK INVITE
	ACK →	→	ACK

SS___XXSS_TIP02	**TIP/TIR reference to: TS 124 508 [24], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**			
TSS reference:	SIP-SIP/SupplementaryServices/TIP.			
Selection criteria:	The originating user subscribes to TIP service. The terminating user subscribes to TIR "temporary mode" default "not restricted".			
Test purpose:	Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request: the originating UE receives, in the 2xx SIP response, a **P-Asserted-Identity** header field with a valid public user identity of the terminating UE.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;			
Comments:	**SIP UA A**	**SUT**		**SIP UA B**
	INVITE 180 Ringing **200 OK INVITE** (P-Asserted-Identity) ACK	→ ← ← →	→ ← ← →	INVITE 180 Ringing 200 OK INVITE ACK

SS___XXSS_TIP03	**TIP/TIR reference to: TS 124 508 [24], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**			
TSS reference:	SIP-SIP/SupplementaryServices/TIP.			
Selection criteria:	The originating user subscribes to TIP service. The terminating user subscribes TIR "temporary mode" default "not restricted".			
Test purpose:	Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request and the Privacy header field is set to "**none**" by the terminating UE in the 2xx SIP response: the originating UE receives, in the 2xx SIP response, a **P-Asserted-Identity** header field with a valid public user identity of the terminating UE.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**none**"			
Comments:	**SIP UA A**	**SUT**		**SIP UA B**
	INVITE 180 Ringing **200 OK INVITE** (P-Asserted-Identity) ACK	→ ← ← →	→ ← ← →	INVITE 180 Ringing 200 OK INVITE ACK

SS___XXSS_TIP04	**TIP/TIR reference to: TS 124 508 [24], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**			
TSS reference:	SIP-SIP/SupplementaryServices/TIP.			
Selection criteria:	The originating user subscribes to TIP service. The terminating user subscribes to TIR "temporary mode" default "restricted".			
Test purpose:	Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request and the Privacy header field is set to "**none**" by the terminating UE in the 2xx SIP response: the originating UE receives, in the 2xx SIP response, a **P-Asserted-Identity** header field with a valid public user identity of the terminating UE.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**none**"			
Comments:	**SIP UA A**	**SUT**		**SIP UA B**
	INVITE 180 Ringing **200 OK INVITE** (P-Asserted-Identity) ACK	→ ← ← →	→ ← ← →	INVITE 180 Ringing 200 OK INVITE ACK

6.2.4 Test purposes for TIR

SS___XXSS_TIR01	**TIP/TIR reference to: TS 124 508 [24], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/TIR.		
Selection criteria:	The originating user subscribes to TIP service. The terminating user subscribes to TIR "temporary mode" default "not restricted".		
Test purpose:	Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request and the Privacy header field is set to "**id**" by the terminating UE in any non-100 SIP response (e.g. 180, 183, 200): the originating UE receives, in any non-100 SIP response (e.g. 180, 183, 200), a Privacy header field is set to "**id**" and no P-Asserted-Identity header field.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**id**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→	→ INVITE
	180 Ringing	←	← 180 Ringing
	200 OK INVITE	←	← 200 OK INVITE
	ACK	→	→ ACK

SS___XXSS_TIR02	**TIP/TIR reference to: TS 124 508 [24], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/TIR.		
Selection criteria:	The originating user subscribes to TIP service. The terminating user subscribes to TIR "temporary mode" default "restricted".		
Test purpose:	Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request and no Privacy header field is inserted by the terminating UE in any non-100 SIP response (e.g. 180, 183, 200): the originating UE receives, in any non-100 SIP response (e.g. 180, 183, 200), a Privacy header field is set to "**id**" and no P-Asserted-Identity header field.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→	→ INVITE
	180 Ringing	←	← 180 Ringing
	200 OK INVITE	←	← 200 OK INVITE
	ACK	→	→ ACK

SS___XXSS_TIR03	**TIP/TIR reference to: TS 124 508 [24], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**		
TSS reference:	SIP-SIP/SupplementaryServices/TIR.		
Selection criteria:	The originating user subscribes to TIP service. The terminating user subscribes to TIR "temporary mode" default "restricted".		
Test purpose:	Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request and the Privacy header field is set to "**id**" by the terminating UE in any non-100 SIP response (e.g. 180, 183, 200): the originating UE receives, in any non-100 SIP response (e.g. 180, 183, 200), a Privacy header field is set to "**id**" and no P-Asserted-Identity header field.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**id**"		
Comments:	**SIP UA A**	**SUT**	**SIP UA B**
	INVITE	→	→ INVITE
	180 Ringing	←	← 180 Ringing
	200 OK INVITE	←	← 200 OK INVITE
	ACK	→	→ ACK

SS___XXSS_TIR04	**TIP/TIR reference to: TS 124 508 [24], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**			
TSS reference:	SIP-SIP/SupplementaryServices/TIR.			
Selection criteria:	The originating user subscribes to TIP service. The terminating user subscribes to TIR "permanent mode".			
Test purpose:	Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request and no Privacy header field is inserted by the terminating UE in any non-100 SIP response (e.g. 180, 183, 200): the originating UE receives, in any non-100 SIP response (e.g. 180, 183, 200), a Privacy header field is set to "**id**" and no P-Asserted-Identity header field.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;			
Comments:	**SIP UA A**		**SUT**	**SIP UA B**
	INVITE	→		→ INVITE
	180 Ringing	←		← 180 Ringing
	200 OK INVITE	←		← 200 OK INVITE
	ACK	→		→ ACK

SS___XXSS_TIR05	**TIP/TIR reference to: TS 124 508 [24], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**			
TSS reference:	SIP-SIP/SupplementaryServices/TIR.			
Selection criteria:	The originating user subscribes to TIP service. The terminating user subscribes to TIR "permanent mode".			
Test purpose:	Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request and the Privacy header field is set to "**id**" by the terminating UE in any non-100 SIP response (e.g. 180, 183, 200): the originating UE receives, in any non-100 SIP response (e.g. 180, 183, 200), a Privacy header field is set to "**id**" and no P-Asserted-Identity header field.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**id**"			
Comments:	**SIP UA A**		**SUT**	**SIP UA B**
	INVITE	→		→ INVITE
	180 Ringing	←		← 180 Ringing
	200 OK INVITE	←		← 200 OK INVITE
	ACK	→		→ ACK

SS___XXSS_TIR06	**TIP/TIR reference to: TS 124 508 [24], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**			
TSS reference:	SIP-SIP/SupplementaryServices/TIP			
Selection criteria:	The originating user subscribes to TIP service. Additionally, the originating user has the "override category". The terminating user subscribes TIR "permanent mode".			
Test purpose:	Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request and no Privacy header field is inserted by the terminating UE in any non-100 SIP response (e.g. 180, 183, 200): The originating UE does not receive a Privacy set to "**id**" in any non-100 SIP response (e.g. 180, 183, 200) and receives, in the 2xx SIP response, a **P-Asserted-Identity** header field with a valid public user identity of the terminating UE.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;			
Comments:	**SIP UA A**		**SUT**	**SIP UA B**
	INVITE	→		→ INVITE
	180 Ringing	←		← 180 Ringing
	200 OK INVITE (P-Asserted-Identity)	←		← 200 OK INVITE
	ACK	→		→ ACK

SS___XXSS_TIR07	**TIP/TIR reference to: TS 124 508 [24], clauses 4.3.2, 4.5.2.1, 4.5.2.4, 4.5.2.12**	
TSS reference:	SIP-SIP/SupplementaryServices/TIP	
Selection criteria:	The originating user subscribes to TIP service. The user subscribes to TIR "permanent mode".	
Test purpose:	Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request and the Privacy header field is set to "**none**" by the terminating UE in any non-100 SIP response (e.g. 180, 183, 200): the originating UE receives, in any non-100 SIP response (e.g. 180, 183, 200), a Privacy header field is set to "**id**" and no P-Asserted-Identity header field.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; Privacy header field is set to "**none**"	
Comments:		

SIP UA A		**SUT**		**SIP UA B**
INVITE	➔		➔	INVITE
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	➔		➔	ACK

6.2.5 Test purposes for Hold

6.2.5.1 Communication Hold with support for UPDATE

SS__XXSSCH01	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE	
Selection criteria:	*Session hold. UPDATE method is used*	
Test purpose:	Ensure that, when the originating UE (user A) sends an UPDATE request containing a SDP with the attribute "a=" sendonly to put the session on hold: • The terminating UE (user B) receives an UPDATE containing a SDP with the attribute "a=" sendonly. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "sendrecv".	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (**sendrecv**)	➔		➔	INVITE (**sendrecv**)
180 Ringing	←		←	180 Ringing
200 OK INVITE (**sendrecv**)	←		←	200 OK INVITE (**sendrecv**)
ACK	➔		➔	ACK
UPDATE (**sendonly**)	➔		➔	**UPDATE (sendonly)**
200 OK UPDATE (**recvonly**)	←		←	**200 OK UPDATE (recvonly)**
BYE	➔		➔	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH02	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE	
Selection criteria:	*Session hold. UPDATE method is used.*	
Test purpose:	Ensure that, when the originating UE (user A) sends an UPDATE request containing a SDP with the attribute "a=" inactive to change the media stream status to inactive: • The terminating UE (user B) receives an UPDATE containing a SDP with the attribute "a=" inactive. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" inactive. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" inactive. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user B (terminating UE).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
		condition: The session was previously put on hold from user B		
UPDATE (sendonly)	←		←	UPDATE(sendonly)
200 OK UPDATE (recvonly)	→		→	200 OK UPDATE (recvonly)
UPDATE (inactive)	→		→	**UPDATE (inactive)**
200 OK UPDATE (inactive)	←		←	**200 OK UPDATE (inactive)**
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

<table>
<tr><td>SS__XXSSCH03</td><td>HOLD reference to:
TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9</td><td></td></tr>
<tr><td>TSS reference:</td><td colspan="2">ServedUser/WithoutAnnounc/WithUPDATE</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">Session hold. UPDATE method is used.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that, when the originating UE (user A) sends an UPDATE request containing a SDP with the attribute "a=" sendrecv to resume the session:
• The terminating UE (user B) receives an UPDATE containing a SDP with the attribute "a=" sendrecv.
• The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv.
• The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv.

Then the originating UE (user A) hang up the session.

NOTE: The sendrecv SDP attribute can be omitted, since sendrecv attribute is the default.</td></tr>
<tr><td>Precondition:</td><td colspan="2">• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures.
• The session was previously put on hold from user A (originating UE).</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Dial string parameters options=PIXIT
TYPE_SDP= PIXIT;</td></tr>
<tr><td colspan="3">Comments:</td></tr>
</table>

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
UPDATE(**sendonly**)	→		→	UPDATE(**sendonly**)
200 OK UPDATE (**recvonly**)	←		←	200 OK UPDATE (**recvonly**)
UPDATE (**sendrecv**)	→		→	**UPDATE** (**sendrecv**)
200 OK UPDATE (**sendrecv**)	←		←	**200 OK UPDATE** (**sendrecv**)
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH04	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE	
Selection criteria:	*Session hold. UPDATE method is used.*	
Test purpose:	Ensure that, when the originating UE (user A) sends an UPDATE request containing a SDP with the attribute "a=" recvonly to resume the media stream status to recvonly: • The terminating UE (user B) receives an UPDATE containing a SDP with the attribute "a=" recvonly. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "inactive" from user A (originating UE).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
UPDATE(sendonly)	←		←	UPDATE(sendonly)
200 OK UPDATE (recvonly)	→		→	200 OK UPDATE (recvonly)
UPDATE (**inactive**)	→		→	UPDATE (**inactive**)
200 OK UPDATE (**inactive**)	←		←	200 OK UPDATE (**inactive**)
UPDATE (**recvonly**)	→		→	**UPDATE** (**recvonly**)
200 OK UPDATE (**sendonly**)	←		←	**200 OK UPDATE** (**sendonly**)
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH05	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE	
Selection criteria:	*Session hold. UPDATE method is used.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an UPDATE request containing a SDP with the attribute "a=" sendonly to put the session on hold: • The originating UE (user A) receives an UPDATE containing a SDP with the attribute "a=" sendonly. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. • The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "sendrecv".	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (**sendrecv**)	→		→	INVITE (**sendrecv**)
180 Ringing	←		←	180 Ringing
200 OK INVITE (**sendrecv**)	←		←	200 OK INVITE (**sendrecv**)
ACK	→		→	ACK
UPDATE (**sendonly**)	←		←	**UPDATE** (**sendonly**)
200 OK UPDATE (**recvonly**)	→		→	**200 OK UPDATE** (**recvonly**)
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH06	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE	
Selection criteria:	*Session hold. UPDATE method is used.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an UPDATE request containing a SDP with the attribute "a=" inactive to change the media stream status to inactive: • The originating UE (user A) receives an UPDATE containing a SDP with the attribute "a=" inactive. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" inactive. • The terminating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" inactive. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user A (originating UE).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
UPDATE (**sendonly**)	→		→	UPDATE (**sendonly**)
200 OK UPDATE (**recvonly**)	←		←	200 OK UPDATE (**recvonly**)
UPDATE (**inactive**)	←		←	**UPDATE** (**inactive**)
200 OK UPDATE (**inactive**)	→		→	**200 OK UPDATE** (**inactive**)
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH07	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE	
Selection criteria:	*Session hold. UPDATE method is used.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an UPDATE request containing a SDP with the attribute "a=" sendrecv to resume the session: • The originating UE (user A) receives an UPDATE containing a SDP with the attribute "a=" sendrecv. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. • The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. Then the originating UE (user A) hang up the session. NOTE: The sendrecv SDP attribute can be omitted, since sendrecv attribute is the default.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user B (terminating UE).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
UPDATE (**sendonly**)	←		←	UPDATE(**sendonly**)
200 OK UPDATE (**recvonly**)	→		→	200 OK UPDATE (**recvonly**)
UPDATE (**sendrecv**)	←		←	**UPDATE** (**sendrecv**)
200 OK UPDATE (**sendrecv**)	→		→	**200 OK UPDATE** (**sendrecv**)
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH08	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE	
Selection criteria:	*Session hold. UPDATE method is used.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an UPDATE request containing a SDP with the attribute "a=" recvonly to resume the media stream status to recvonly: • The originating UE (user A) receives an UPDATE containing a SDP with the attribute "a=" recvonly. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. • The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "inactive" from user B (terminating UE).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
UPDATE (sendonly)	→		→	UPDATE (sendonly)
200 OK UPDATE (recvonly)	←		←	200 OK UPDATE (recvonly)
UPDATE (**inactive**)	←		←	UPDATE (**inactive**)
200 OK UPDATE (**inactive**)	→		→	200 OK UPDATE (**inactive**)
UPDATE(**recvonly**)	←		←	**UPDATE**(**recvonly**)
200 OK UPDATE (**sendonly**)	→		→	**200 OK UPDATE** (**sendonly**)
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

6.2.5.2 Communication Hold without support for UPDATE

SS__XXSSCH09	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE	
Selection criteria:	*Session hold. INVITE method is used.*	
Test purpose:	Ensure that, when the originating UE (user A) sends an INVITE request containing a SDP with the attribute "a=" sendonly to put the session on hold: • The terminating UE (user B) receives an INVITE containing a SDP with the attribute "a=" sendonly. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "sendrecv".	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (**sendrecv**)	→		→	INVITE (**sendrecv**)
180 Ringing	←		←	180 Ringing
200 OK INVITE (**sendrecv**)	←		←	200 OK INVITE (**sendrecv**)
ACK	→		→	ACK
INVITE (**sendonly**)	→		→	**INVITE** (**sendonly**)
200 OK INVITE (**recvonly**)	←		←	**200 OK INVITE**(**recvonly**)
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 10	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE	
Selection criteria:	*Session hold. INVITE method is used.*	
Test purpose:	Ensure that, when the originating UE (user A) sends an INVITE request containing a SDP with the attribute "a=" inactive to change the media stream status to inactive: • The terminating UE (user B) receives an INVITE containing a SDP with the attribute "a=" inactive. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" inactive. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" inactive. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user B (terminating UE).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
INVITE (**sendonly**)	←		←	INVITE (**sendonly**)
200 OK INVITE (**recvonly**)	→		→	200 OK INVITE (**recvonly**)
ACK	←		←	ACK
INVITE (**inactive**)	→		→	**INVITE** (**inactive**)
200 OK INVITE (**inactive**)	←		←	**200 OK INVITE** (**inactive**)
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 11	**HOLD reference to:** **TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE	
Selection criteria:	*Session hold. INVITE method is used.*	
Test purpose:	Ensure that, when the originating UE (user A) sends an INVITE request containing a SDP with the attribute "a=" sendrecv to resume the session: • The terminating UE (user B) receives an INVITE containing a SDP with the attribute "a=" sendrecv. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. Then the originating UE (user A) hang up the session. NOTE: The sendrecv SDP attribute can be omitted, since sendrecv attribute is the default.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user A (originating UE).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
INVITE (**sendonly**)	→		→	INVITE (**sendonly**)
200 OK INVITE (**recvonly**)	←		←	200 OK INVITE(**recvonly**)
ACK	→		→	ACK
INVITE (**sendrecv**)	→		→	**INVITE** (**sendrecv**)
200 OK INVITE (**sendrecv**)	←		←	**200 OK INVITE** (**sendrecv**)
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

<table>
<tr><td>SS__XXSSCH 12</td><td>HOLD reference to:
TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9</td><td></td></tr>
<tr><td>TSS reference:</td><td colspan="2">ServedUser/WithoutAnnounc/WithoutUPDATE</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">Session hold. INVITE method is used.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that, when the originating UE (user A) sends an INVITE request containing a SDP with the attribute "a=" recvonly to resume the media stream status to recvonly:
The terminating UE (user B) receives an INVITE containing a SDP with the attribute "a=" recvonly.
The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendonly.
The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendonly.
Then the originating UE (user A) hang up the session.</td></tr>
<tr><td>Precondition:</td><td colspan="2">A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures.
The media stream was previously set to "inactive" from user A (originating UE).</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Dial string parameters options=PIXIT
TYPE_SDP= PIXIT;</td></tr>
<tr><td colspan="3">Comments:</td></tr>
</table>

SIP UA A		SUT		SIP UA B
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
INVITE(sendonly)	←		←	INVITE(sendonly)
200 OK INVITE (recvonly)	→		→	200 OK INVITE(recvonly)
ACK	←		←	ACK
INVITE (**inactive**)	→		→	INVITE (**inactive**)
200 OK INVITE (**inactive**)	←		←	200 OK INVITE (**inactive**)
ACK	→		→	ACK
INVITE (**recvonly**)	→		→	**INVITE** (**recvonly**)
200 OK INVITE (**sendonly**)	←		←	**200 OK INVITE** (**sendonly**)
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 13	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE
Selection criteria:	*Session hold. INVITE method is used.*
Test purpose:	Ensure that, when the terminating UE (user B) sends an INVITE request containing a SDP with the attribute "a=" sendonly to put the session on hold: • The originating UE (user A) receives an INVITE containing a SDP with the attribute "a=" sendonly. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. • The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. Then the originating UE (user A) hang up the session.
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "sendrecv".
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (**sendrecv**)	→		→	INVITE (**sendrecv**)
180 Ringing	←		←	180 Ringing
200 OK INVITE (**sendrecv**)	←		←	200 OK INVITE (**sendrecv**)
ACK	→		→	ACK
INVITE (**sendonly**)	←		←	**INVITE** (**sendonly**)
200 OK INVITE (**recvonly**)	→		→	**200 OK INVITE** (**recvonly**)
ACK	←		←	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

<table>
<tr><td>SS__XXSSCH 14</td><td>HOLD reference to:
TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9</td><td></td></tr>
<tr><td>TSS reference:</td><td colspan="2">ServedUser/WithoutAnnounc/WithoutUPDATE</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">Session hold. INVITE method is used.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that, when the terminating UE (user B) sends an INVITE request containing a SDP with the attribute "a=" inactive to change the media stream status to inactive:
• The originating UE (user A) receives an INVITE containing a SDP with the attribute "a=" inactive.
• The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" inactive.
• The terminating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" inactive.

Then the originating UE (user A) hang up the session.</td></tr>
<tr><td>Precondition:</td><td colspan="2">• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures.
• The session was previously put on hold from user A (originating UE).</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Dial string parameters options=PIXIT
TYPE_SDP= PIXIT;</td></tr>
<tr><td colspan="3">Comments:</td></tr>
</table>

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
INVITE (**sendonly**)	→		→	INVITE (**sendonly**)
200 OK INVITE (**recvonly**)	←		←	200 OK INVITE (**recvonly**)
ACK	→		→	ACK
INVITE(**inactive**)	←		←	**INVITE** (**inactive**)
200 OK INVITE (**inactive**)	→		→	**200 OK INVITE** (**inactive**)
ACK	←		←	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 15	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE	
Selection criteria:	*Session hold. INVITE method is used.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an INVITE request containing a SDP with the attribute "a=" sendrecv to resume the session: • The originating UE (user A) receives an INVITE containing a SDP with the attribute "a=" sendrecv. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. • The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. Then the originating UE (user A) hang up the session. NOTE: The sendrecv SDP attribute can be omitted, since sendrecv attribute is the default.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user B (terminating UE).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
INVITE (**sendonly**)	←		←	INVITE(**sendonly**)
200 OK INVITE (**recvonly**)	→		→	200 OK INVITE(**recvonly**)
ACK	←		←	ACK
INVITE (**sendrecv**)	←		←	**INVITE** (**sendrecv**)
200 OK INVITE (**sendrecv**)	→		→	**200 OK INVITE** (**sendrecv**)
ACK	←		←	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 16	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE	
Selection criteria:	*Session hold. INVITE method is used.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an INVITE request containing a SDP with the attribute "a=" recvonly to resume the media stream status to recvonly: • The originating UE (user A) receives an INVITE containing a SDP with the attribute "a=" recvonly. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. • The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "inactive" from user B (terminating UE).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	
Comments:		

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
INVITE(sendonly)	→		→	INVITE(sendonly)
200 OK INVITE (recvonly)	←		←	200 OK INVITE(recvonly)
ACK	→		→	ACK
INVITE (**inactive**)	←		←	INVITE (**inactive**)
200 OK INVITE (**inactive**)	→		→	200 OK INVITE(**inactive**)
ACK	←		←	ACK
INVITE (**recvonly**)	←		←	**INVITE (recvonly)**
200 OK INVITE (**sendonly**)	→		→	**200 OK INVITE** (**sendonly**)
ACK	←		←	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

6.2.5.3 Communication with announcements

6.2.5.3.1 Communication Hold with support for UPDATE

SS__XXSSCH17	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithAnnounc/WithUPDATE	
Selection criteria:	*The remote user is put on hold, an announcement starts to the held user. The UPDATE method is used.*	
Test purpose:	Ensure that, when the originating UE (user A) sends an UPDATE request containing a SDP with the attribute "a=" sendonly to put the session on hold: • The terminating UE (user B) receives an UPDATE containing a SDP with the attribute "a=" sendonly. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. • An announcement is played to the terminating UE (user B). Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "sendrecv".	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
UPDATE (**sendonly**)	→		→	**UPDATE** (**sendonly**)
200 OK UPDATE (**recvonly**)	←		←	**200 OK UPDATE** (**recvonly**)
		Announcement to UE B		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH18	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithAnnounc/WithUPDATE	
Selection criteria:	The announcement is stopped after the held user puts the media stream on hold. The UPDATE method is used.	
Test purpose:	Ensure that, when the originating UE (user A) sends an UPDATE request containing a SDP with the attribute "a=" inactive to change the media stream status to inactive: • The terminating UE (user B) receives an UPDATE containing a SDP with the attribute "a=" inactive. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" inactive. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" inactive. • The announcement to the originating UE (user A) is stopped. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user B (terminating UE). • An announcement is played to the originating UE (user A).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	
Comments:		

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
UPDATE (**sendonly**)	←		←	UPDATE (**sendonly**)
200 OK UPDATE (**recvonly**)	→		→	200 OK UPDATE (**recvonly**)
		Announcement to UE A		
UPDATE (**inactive**)	→		→	**UPDATE** (**inactive**)
200 OK UPDATE (**inactive**)	←		←	**200 OK UPDATE** (**inactive**)
		Media stream is stopped		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH19	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithAnnounc/WithUPDATE	
Selection criteria:	The announcement is stopped after retrieve.	
Test purpose:	Ensure that, when the originating UE (user A) sends an UPDATE request containing a SDP with the attribute "a=" sendrecv to resume the session: • The terminating UE (user B) receives an UPDATE containing a SDP with the attribute "a=" sendrecv. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. • The announcement to the terminating UE (user B) is stopped. Then the originating UE (user A) hang up the session. NOTE: The sendrecv SDP attribute can be omitted, since sendrecv attribute is the default.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user A (originating UE). • An announcement is played to the terminating UE (user B).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
UPDATE (**sendonly**)	→		→	UPDATE (**sendonly**)
200 OK UPDATE (**recvonly**)	←		←	200 OK UPDATE (**recvonly**)
		Announcement to UE B		
UPDATE (**sendrecv**)	→		→	**UPDATE** (**sendrecv**)
200 OK UPDATE (**sendrecv**)	←		←	**200 OK UPDATE** (**sendrecv**)
		Conversation		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

<table>
<tr><td>SS__XXSSCH20</td><td>HOLD reference to:
TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9</td><td></td></tr>
<tr><td>TSS reference:</td><td colspan="2">ServedUser/WithAnnounc/WithUPDATE</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">Announcement is started to user B when user B retrieves the connection.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that, when the originating UE (user A) sends an UPDATE request containing a SDP with the attribute "a=" recvonly to resume the media stream status to recvonly:
The terminating UE (user B) receives an UPDATE containing a SDP with the attribute "a=" recvonly.
The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendonly.
The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendonly.
An announcement is played to the originating UE (user A).
Then the originating UE (user A) hang up the session.</td></tr>
<tr><td>Precondition:</td><td colspan="2">A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures.
The media stream was previously set to "inactive" from user A (originating UE).
The announcement to the originating UE (user A) is stopped.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Dial string parameters options=PIXIT
TYPE_SDP= PIXIT;</td></tr>
<tr><td colspan="3">Comments:</td></tr>
</table>

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
		Conversation		
UPDATE (sendonly)	←		←	UPDATE (sendonly)
200 OK UPDATE (recvonly)	→		→	200 OK UPDATE (recvonly)
		Announcement to UE A		
UPDATE (**inactive**)	→		→	UPDATE (**inactive**)
200 OK UPDATE (**inactive**)	←		←	200 OK UPDATE (**inactive**)
		Media stream is stopped		
UPDATE (**recvonly**)	→		→	**UPDATE** (**recvonly**)
200 OK UPDATE (**sendonly**)	←		←	**200 OK UPDATE** (**sendonly**)
		Announcement to UE A		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH21	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**
TSS reference:	ServedUser/WithAnnounc/WithUPDATE
Selection criteria:	*The remote user is put on hold, an announcement starts to the held user. The UPDATE method is used.*
Test purpose:	Ensure that, when the terminating UE (user B) sends an UPDATE request containing a SDP with the attribute "a=" sendonly to put the session on hold: • The originating UE (user A) receives an UPDATE containing a SDP with the attribute "a=" sendonly. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. • The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. • An announcement is played to the terminating UE (user B). Then the originating UE (user A) hang up the session.
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "sendrecv".
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (**sendrecv**)	→		→	INVITE (**sendrecv**)
180 Ringing	←		←	180 Ringing
200 OK INVITE (**sendrecv**)	←		←	200 OK INVITE (**sendrecv**)
ACK	→		→	ACK
		Conversation		
UPDATE (**sendonly**)	←		←	**UPDATE** (**sendonly**)
200 OK UPDATE (**recvonly**)	→		→	**200 OK UPDATE** (**recvonly**)
		Announcement to UE A		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH22	HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9	
TSS reference:	ServedUser/WithAnnounc/WithUPDATE	
Selection criteria:	*The announcement is stopped after the held user puts the media stream on hold. The UPDATE method is used.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an UPDATE request containing a SDP with the attribute "a=" inactive to change the media stream status to inactive: • The originating UE (user A) receives an UPDATE containing a SDP with the attribute "a=" inactive. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" inactive. • The terminating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" inactive. • The announcement to the terminating UE (user B) is stopped. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user A (originating UE). • An announcement is played to the terminating UE (user B).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		SUT		SIP UA B
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
		Conversation		
UPDATE (**sendonly**)	→		→	UPDATE(**sendonly**)
200 OK UPDATE (**recvonly**)	←		←	200 OK UPDATE (**recvonly**)
		Announcement to UE B		
UPDATE (**inactive**)	←		←	**UPDATE** (**inactive**)
200 OK UPDATE (**inactive**)	→		→	**200 OK UPDATE** (**inactive**)
		Media stream is stopped		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH23	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithAnnounc/WithUPDATE	
Selection criteria:	*Announcement is stopped after retrieve.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an UPDATE request containing a SDP with the attribute "a=" sendrecv to resume the session: • The originating UE (user A) receives an UPDATE containing a SDP with the attribute "a=" sendrecv. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. • The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. • The announcement to the originating UE (user A) is stopped. Then the originating UE (user A) hang up the session. NOTE: The sendrecv SDP attribute can be omitted, since sendrecv attribute is the default.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user B (terminating UE). • An announcement is played to the originating UE (user A).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
		Conversation		
UPDATE (**sendonly**)	←		←	UPDATE (**sendonly**)
200 OK UPDATE (**recvonly**)	→		→	200 OK UPDATE (**recvonly**)
		Announcement to UE A		
UPDATE (**sendrecv**)	←		←	**UPDATE** (**sendrecv**)
200 OK UPDATE (**sendrecv**)	→		→	**200 OK UPDATE** (**sendrecv**)
		Conversation		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH24	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithAnnounc/WithUPDATE	
Selection criteria:	*Announcement is started to user B when user B retrieves the connection.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an UPDATE request containing a SDP with the attribute "a=" recvonly to resume the media stream status to recvonly: • The originating UE (user A) receives an UPDATE containing a SDP with the attribute "a=" recvonly. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. • The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. • An announcement is played to the terminating UE (user B). Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "inactive" from user B (terminating UE). • The announcement to the terminating UE (user B) is stopped.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
		Conversation		
UPDATE (sendonly)	→		→	UPDATE (sendonly)
200 OK UPDATE (recvonly)	←		←	200 OK UPDATE (recvonly)
		Announcement to UE B		
UPDATE (**inactive**)	←		←	UPDATE (**inactive**)
200 OK UPDATE (**inactive**)	→		→	200 OK UPDATE (**inactive**)
		Media stream is stopped		
UPDATE (**recvonly**)	←		←	**UPDATE** (**recvonly**)
200 OK UPDATE (**sendonly**)	→		→	**200 OK UPDATE** (**sendonly**)
		Announcement to UE B		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

6.2.5.3.2 Communication Hold without support for UPDATE

SS__XXSSCH25	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithAnnounc/WithoutUPDATE	
Selection criteria:	*The remote user is put on hold, an announcement starts to the held user. The INVITE method is used.*	
Test purpose:	Ensure that, when the originating UE (user A) sends an INVITE request containing a SDP with the attribute "a=" sendonly to put the session on hold: • The terminating UE (user B) receives an INVITE containing a SDP with the attribute "a=" sendonly. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" recvonly. • An announcement is played to the terminating UE (user B). Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "sendrecv".	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (**sendrecv**)	→		→	INVITE (**sendrecv**)
180 Ringing	←		←	180 Ringing
200 OK INVITE (**sendrecv**)	←		←	200 OK INVITE (**sendrecv**)
ACK	→		→	ACK
		Conversation		
INVITE (**sendonly**)	→		→	**INVITE** (**sendonly**)
200 OK INVITE (**recvonly**)	←		←	**200 OK INVITE** (**recvonly**)
ACK	→		→	ACK
		Announcement to UE B		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH26	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithAnnounc/WithoutUPDATE	
Selection criteria:	*The announcement is stopped after the held user puts the media stream on hold. The INVITE method is used.*	
Test purpose:	Ensure that, when the originating UE (user A) sends an INVITE request containing a SDP with the attribute "a=" inactive to change the media stream status to inactive: • The terminating UE (user B) receives an INVITE containing a SDP with the attribute "a=" inactive. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" inactive. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" inactive. • The announcement to the originating UE (user A) is stopped. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user B (terminating UE). • An announcement is played to the originating UE (user A).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	
Comments:		

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
		Conversation		
INVITE (**sendonly**)	←		←	INVITE (**sendonly**)
200 OK INVITE (**recvonly**)	→		→	200 OK INVITE (**recvonly**)
ACK	←		←	ACK
		Announcement to UE A		
INVITE (**inactive**)	→		→	**INVITE** (**inactive**)
200 OK INVITE (inactive)	←		←	**200 OK INVITE** (**inactive**)
ACK	→		→	ACK
		Media stream is stopped		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

<table>
<tr><td>SS__XXSSCH27</td><td>HOLD reference to:
TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9</td><td></td></tr>
<tr><td>TSS reference:</td><td colspan="2">ServedUser/WithAnnounc/WithoutUPDATE</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">Announcement is stopped after retrieve. The INVITE method is used.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that, when the originating UE (user A) sends an INVITE request containing a SDP with the attribute "a=" sendrecv to resume the session:
• The terminating UE (user B) receives an INVITE containing a SDP with the attribute "a=" sendrecv.
• The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv.
• The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv.
• The announcement to the terminating UE (user B) is stopped.

Then the originating UE (user A) hang up the session.

NOTE: The sendrecv SDP attribute can be omitted, since sendrecv attribute is the default.</td></tr>
<tr><td>Precondition:</td><td colspan="2">• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures.
• The session was previously put on hold from user A (originating UE).
• An announcement is played to the terminating UE (user B).</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Dial string parameters options=PIXIT
TYPE_SDP= PIXIT;</td></tr>
<tr><td colspan="3">Comments:
<table>
<tr><th>SIP UA A</th><th></th><th>SUT</th><th></th><th>SIP UA B</th></tr>
<tr><td>INVITE (sendrecv)</td><td>→</td><td></td><td>→</td><td>INVITE (sendrecv)</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>200 OK INVITE (sendrecv)</td><td>←</td><td></td><td>←</td><td>200 OK INVITE (sendrecv)</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>INVITE (sendonly)</td><td>→</td><td></td><td>→</td><td>INVITE (sendonly)</td></tr>
<tr><td>200 OK INVITE (recvonly)</td><td>←</td><td></td><td>←</td><td>200 OK INVITE (recvonly)</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Announcement to UE B</td><td></td><td></td></tr>
<tr><td>INVITE (sendrecv)</td><td>→</td><td></td><td>→</td><td>INVITE (sendrecv)</td></tr>
<tr><td>200 OK INVITE (sendrecv)</td><td>←</td><td></td><td>←</td><td>200 OK INVITE (sendrecv)</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table></td></tr>
</table>

SS__XXSSCH28	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithAnnounc/WithoutUPDATE	
Selection criteria:	*Announcement is started to user B when user B retrieves the connection. The INVITE method is used.*	
Test purpose:	Ensure that, when the originating UE (user A) sends an INVITE request containing a SDP with the attribute "a=" recvonly to resume the media stream status to recvonly: • The terminating UE (user B) receives an INVITE containing a SDP with the attribute "a=" recvonly. • The terminating UE (user B) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. • The originating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. • An announcement is played to the originating UE (user A). Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "inactive" from user A (originating UE). • The announcement to the originating UE (user A) is stopped.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
		Conversation		
INVITE(sendonly)	←		←	INVITE(sendonly)
200 OK INVITE (recvonly)	→		→	200 OK INVITE(recvonly)
ACK	←		←	ACK
		Announcement to UE A		
INVITE (inactive)	→		→	INVITE(inactive)
200 OK INVITE (inactive)	←		←	200 OK INVITE(inactive)
ACK	→		→	ACK
		Media stream is stopped		
INVITE (recvonly)	→		→	INVITE (**recvonly**)
200 OK INVITE (sendonly)	←		←	200 OK INVITE (**sendonly**)
ACK	→		→	ACK
		Announcement to UE A		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

<table>
<tr><th>SS__XXSSCH29</th><th>HOLD reference to:
TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9</th><th></th></tr>
<tr><td>TSS reference:</td><td colspan="2">ServedUser/WithAnnounc/WithoutUPDATE</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">The remote user is put on hold, an announcement starts to the held user. The INVITE method is used.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that, when the terminating UE (user B) sends an INVITE request containing a SDP with the attribute "a=" sendonly to put the session on hold:
• The originating UE (user A) receives an INVITE containing a SDP with the attribute "a=" sendonly.
• The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" recvonly.
• The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" recvonly.
• An announcement is played to the terminating UE (user B).

Then the originating UE (user A) hang up the session.</td></tr>
<tr><td>Precondition:</td><td colspan="2">• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures.
• The media stream was previously set to "sendrecv".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Dial string parameters options=PIXIT
TYPE_SDP= PIXIT;</td></tr>
<tr><td colspan="3">Comments:</td></tr>
</table>

SIP UA A		SUT		SIP UA B
INVITE (**sendrecv**)	→		→	INVITE (**sendrecv**)
180 Ringing	←		←	180 Ringing
200 OK INVITE (**sendrecv**)	←		←	200 OK INVITE (**sendrecv**)
ACK	→		→	ACK
		Conversation		
INVITE (sendonly)	←		←	INVITE (**sendonly**)
200 OK INVITE (recvonly)	→		→	200 OK INVITE (**recvonly**)
ACK	→		→	ACK
		Announcement to UE A		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH30	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithAnnounc/WithoutUPDATE	
Selection criteria:	*The announcement is stopped after the held user puts the media stream on hold. The INVITE method is used.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an INVITE request containing a SDP with the attribute "a=" inactive to change the media stream status to inactive: • The originating UE (user A) receives an INVITE containing a SDP with the attribute "a=" inactive. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" inactive. • The terminating UE (user A) receives a 200 OK SIP response containing a SDP with the attribute "a=" inactive. • The announcement to the terminating UE (user B) is stopped. Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user A (originating UE). • An announcement is played to the terminating UE (user B).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
		Conversation		
INVITE (**sendonly**)	→		→	INVITE (**sendonly**)
200 OK INVITE (**recvonly**)	←		←	200 OK INVITE (**recvonly**)
ACK	→		→	ACK
		Announcement to UE B		
INVITE (**inactive**)	←		←	**INVITE** (**inactive**)
200 OK INVITE (**inactive**)	→		→	**200 OK INVITE** (**inactive**)
ACK	→		→	ACK
		Media stream is stopped		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH31	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithAnnounc/WithoutUPDATE	
Selection criteria:	*Announcement is stopped after retrieve. The INVITE method id used.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an INVITE request containing a SDP with the attribute "a=" sendrecv to resume the session: • The originating UE (user A) receives an INVITE containing a SDP with the attribute "a=" sendrecv. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. • The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendrecv. • The announcement to the originating UE (user A) is stopped. Then the originating UE (user A) hang up the session. NOTE: The sendrecv SDP attribute can be omitted, since sendrecv attribute is the default.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The session was previously put on hold from user B (terminating UE). • An announcement is played to the originating UE (user A).	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (sendrecv)	→		→	INVITE (sendrecv)
180 Ringing	←		←	180 Ringing
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (sendrecv)
ACK	→		→	ACK
		Conversation		
INVITE (**sendonly**)	←		←	INVITE (**sendonly**)
200 OK INVITE (**recvonly**)	→		→	200 OK INVITE (**recvonly**)
ACK	←		←	ACK
		Announcement to UE A		
INVITE (**sendrecv**)	←		←	**INVITE** (**sendrecv**)
200 OK INVITE (**sendrecv**)	→		→	**200 OK INVITE** (**sendrecv**)
ACK	←		←	ACK
		Conversation		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH32	**HOLD reference to: TS 124 410 [15], clauses 4.5.2.1, 4.5.2.4, 4.5.2.9**	
TSS reference:	ServedUser/WithAnnounc/WithoutUPDATE	
Selection criteria:	*Announcement is started to user B when user B retrieves the connection. The INVITE method id used.*	
Test purpose:	Ensure that, when the terminating UE (user B) sends an INVITE request containing a SDP with the attribute "a=" recvonly to resume the media stream status to recvonly: • The originating UE (user A) receives an INVITE containing a SDP with the attribute "a=" recvonly. • The originating UE (user A) sends a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. • The terminating UE (user B) receives a 200 OK SIP response containing a SDP with the attribute "a=" sendonly. • An announcement is played to the terminating UE (user B). Then the originating UE (user A) hang up the session.	
Precondition:	• A session was established between user A (originating UE) and user B (terminating UE) according to the "basic Call" procedures. • The media stream was previously set to "inactive" from user B (terminating UE). • The announcement to the terminating UE (user B) is stopped.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	

Comments:

SIP UA A		**SUT**		**SIP UA B**
INVITE (**sendrecv**)	→		→	INVITE
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
		Conversation		
INVITE(sendonly)	→		→	INVITE(sendonly)
200 OK INVITE (recvonly)	←		←	200 OK INVITE(recvonly)
ACK	→		→	ACK
		Announcement to UE B		
INVITE (inactive)	←		←	INVITE(inactive)
200 OK INVITE (inactive)	→		→	200 OK INVITE(inactive)
ACK	←		←	ACK
		Media stream is stopped		
INVITE (recvonly)	←		←	INVITE (**recvonly**)
200 OK INVITE (sendonly)	→		→	200 OK INVITE (**sendonly**)
ACK	←		←	ACK
		Announcement to UE B		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

6.2.6 Test purposes for Communication Diversion

The configuration lines in this clause contain only the subscription options to the communication diversion service that are relevant for the test purpose. Subscription options not mentioned can take any value.

6.2.6.1 CFU

SSS__XXSSCFU01	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFU	
Configuration:	The **user B** has subscribed to CFU, CDIVN is not activated **Subscription options:** Served user receives indication that a communication has been forwarded = **Yes**	
Selection criteria:	CFU supported.	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B receives a MESSAGE request indicating the call diversion.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	
Comments:		

SIP UA A	**SUT**	**SIP UA B**	**SIP UA C**
INVITE	→		
	Communication diversion is performed		
181 Call Is Being Forwarded (optional)	←		→ INVITE
		→ MESSAGE	
		← 200 OK MESSAGE	
180 Ringing	←		← 180 Ringing
200 OK INVITE	←		← 200 OK INVITE
ACK	→		→ ACK
	Communication		
BYE	→		→ BYE
200 OK BYE	←		← 200 OK BYE

SSS__XXSSCFU02	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.5.1**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFU	
Configuration:	The **user B** has subscribed to CFU and CDIVN	
Selection criteria:	CFU and CDIVN supported.	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B, having activated the CDIVN service, receives a NOTIFY request indicating the call diversion.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	
Comments:		

SIP UA A	**SUT**	**SIP UA B**	**SIP UA C**
	Start Activation CDIVN		
		← SUBSCRIBE	
		→ 200 OK SUBSCRIBE	
		→ NOTIFY	
		← 200 OK NOTIFY	
	End Activation CDIVN		
INVITE	→		
	Communication diversion is performed		
181 Call Is Being Forwarded (optional)	←		→ INVITE
		→ NOTIFY	
		← 200 OK NOTIFY	
180 Ringing	←		← 180 Ringing
200 OK INVITE	←		← 200 OK INVITE
ACK	→		→ ACK
	Communication		
BYE	→		→ BYE
200 OK BYE	←		← 200 OK BYE

SSS__XXSSCFU03	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**		
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFU		
Configuration:	The **user B** has subscribed to CFU **Subscription options:** Originating user receives notification that his communication has been diverted = **No**		
Selection criteria:	CFU supported.		
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded message.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT		
Comments:			
SIP UA A		**SUT**	**SIP UA C**
INVITE	→	→	INVITE
180 Ringing	←	←	180 Ringing
200 OK INVITE	←	←	200 OK INVITE
ACK	→	→	ACK
BYE	→	→	BYE
200 OK BYE	←	←	200 OK BYE

SSS__XXSSCFU04	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**		
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFU		
Configuration:	The **user B** has subscribed to CFU and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **No** Served user allows the presentation of his/her URI to originating user in diversion notification = **No**		
Selection criteria:	CFU supported.		
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a Privacy header with value "id" and not containing a P-Asserted-Identity indicating the URI of user B and not containing a History-Info header (with CDIV related cause value) indicating the URI of user B or user A.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT		
Comments:			
SIP UA A		**SUT**	**SIP UA C**
INVITE	→	→	INVITE
181 Call Is Being Forwarded	←		
180 Ringing	←	←	180 Ringing
200 OK INVITE	←	←	200 OK INVITE
ACK	→	→	ACK
BYE	→	→	BYE
200 OK BYE	←	←	200 OK BYE

SSS__XXSSCFU05	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFU
Configuration:	The **user B** has subscribed to CFU and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **Yes** Served user allows the presentation of his/her URI to originating user in diversion notification = **Yes**
Selection criteria:	CFU supported.
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a P-Asserted-Identity indicating the URI of user B and containing a History-Info header including a first entry with the hi-targeted-to-URI of user B, index = 1, cause param = 302 and including a second entry with the hi-targeted-to-URI of user C, index = 1.1
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA C**
INVITE	→		→	INVITE
181 Call Is Being Forwarded	←			
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SSS__XXSSCFU06	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.2.2**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFU
Configuration:	The **user B** has subscribed to CFU and has not activated OIR **Subscription options:** Served user allows the presentation of his/her URI to diverted-to user = **Yes**
Selection criteria:	CFU supported.
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that User C receives an INVITE message containing a History-Info header including an entry (with CDIV related cause value) with the hi-targeted-to-URI of user B.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA C**
INVITE	→		→	INVITE
181 Call Is Being Forwarded (optional)	←			
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

6.2.6.2 CFB

6.2.6.2.1 NDUB

SSS__XXSSCFB01	**CDIV reference to:** **TS 124 504 [12], clause 4.5.2.6.5**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB
Configuration:	The **user B** has subscribed to CFB, CDIVN is not activated **Subscription options:** Served user receives indication that a communication has been forwarded = **Yes** The **user B** has **not** subscribed to CW
Selection criteria:	CFB supported, NDUB status can be achieved for user B.
Test purpose:	Ensure that when user A calls user B which is network determined user busy (NDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B receives a MESSAGE request indicating the call diversion.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A	**SUT**		**SIP UA B**		**SIP UA C**
UA B enters NDUB condition (e.g. by establishing a communication)					
INVITE	→				
Communication diversion is performed					
181 Call Is Being Forwarded (optional)	←			→	INVITE
		→	MESSAGE		
		←	200 OK MESSAGE		
180 Ringing	←			←	180 Ringing
200 OK INVITE	←			←	200 OK INVITE
ACK	→			→	ACK
Communication					
BYE	→			→	BYE
200 OK BYE	←			←	200 OK BYE

SSS__XXSSCFB02	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.5.1**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB
Configuration:	The **user B** has subscribed to CFB and CDIVN The **user B** has **not** subscribed to CW
Selection criteria:	CFB and CDIVN supported, NDUB status can be achieved for user B.
Test purpose:	Ensure that when user A calls user B which is network determined user busy (NDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B, having activated the CDIVN service, receives a NOTIFY request indicating the call diversion.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
		Activation CDIVN				
			←	SUBSCRIBE		
			→	200 OK SUBSCRIBE		
			→	NOTIFY		
			←	200 OK NOTIFY		
		UA B enters NDUB condition (e.g. by establishing a communication)				
INVITE	→					
		Communication diversion is performed				
181 Call Is Being Forwarded (optional)	←				→	INVITE
			→	NOTIFY		
			←	200 OK NOTIFY		
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

SSS__XXSSCFB03	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB
Configuration:	The **user B** has subscribed to CFB **Subscription options:** Originating user receives notification that his communication has been diverted = **No** The **user B** has **not** subscribed to CW
Selection criteria:	CFB supported, NDUB status can be achieved for user B.
Test purpose:	Ensure that when user A calls user B which is network determined user busy (NDUB), the call is forwarded to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded message.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA C**
		UA B enters NDUB condition (e.g. by establishing a communication)		
INVITE	→		→	INVITE
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SSS__XXSSCFB04	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB
Configuration:	The **user B** has subscribed to CFB and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **No** Served user allows the presentation of his/her URI to originating user in diversion notification = **No** The **user B** has **not** subscribed to CW
Selection criteria:	CFB supported, NDUB status can be achieved for user B.
Test purpose:	Ensure that when user A calls user B which is network determined user busy (NDUB), the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a Privacy header with value "id" and not containing a P-Asserted-Identity indicating the URI of user B and not containing a History-Info header (with CDIV related cause value) indicating the URI of user B or user A.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A	**SUT**		**SIP UA C**
UA B enters NDUB condition (e.g. by establishing a communication)			
INVITE	→	→	INVITE
181 Call Is Being Forwarded	←		
180 Ringing	←	←	180 Ringing
200 OK INVITE	←	←	200 OK INVITE
ACK	→	→	ACK
BYE	→	→	BYE
200 OK BYE	←	←	200 OK BYE

SSS__XXSSCFB05	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB
Configuration:	The **user B** has subscribed to CFB and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **Yes** Served user allows the presentation of his/her URI to originating user in diversion notification = **Yes** The **user B** has **not** subscribed to CW
Selection criteria:	CFB supported, NDUB status can be achieved for user B.
Test purpose:	Ensure that when user A calls user B which is network determined user busy (NDUB), the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a P-Asserted-Identity indicating the URI of user B and containing a History-Info header including a first entry with the hi-targeted-to-URI of user B, index = 1, cause param = 486 and including a second entry with the hi-targeted-to-URI of user C, index = 1.1.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A	**SUT**		**SIP UA C**
UA B enters NDUB condition (e.g. by establishing a communication)			
INVITE	→	→	INVITE
181 Call Is Being Forwarded	←		
180 Ringing	←	←	180 Ringing
200 OK INVITE	←	←	200 OK INVITE
ACK	→	→	ACK
BYE	→	→	BYE
200 OK BYE	←	←	200 OK BYE

SSS__XXSSCFB06	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.2.2**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB	
Configuration:	The **user B** has subscribed to CFB and has not activated OIR **Subscription options:** Served user allows the presentation of his/her URI to diverted-to user = **Yes** The **user B** has **not** subscribed to CW	
Selection criteria:	CFB supported, NDUB status can be achieved for user B.	
Test purpose:	Ensure that when user A calls user B which is network determined user busy (NDUB), the call is forwarded to user C. Ensure that User C receives an INVITE message containing a History-Info header including an entry (with CDIV related cause value) with the hi-targeted-to-URI of user B.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	
Comments:		

SIP UA A		**SUT**		**SIP UA C**
		UA B enters NDUB condition (e.g. by establishing a communication)		
INVITE	→		→	INVITE
181 Call Is Being Forwarded (optional)	←			
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

6.2.6.2.2 UDUB

SSS__XXSSCFB07	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB	
Configuration:	The **user B** has subscribed to CFB, CDIVN is not activated **Subscription options:** Served user receives indication that a communication has been forwarded = **Yes**	
Selection criteria:	CFB supported.	
Test purpose:	Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B receives a MESSAGE request indicating the call diversion.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	
Comments:		

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
			←	486 Busy Here		
			→	ACK		
		Communication diversion is performed				
181 Call Is Being Forwarded (optional)	←				→	INVITE
			→	MESSAGE		
			←	200 OK MESSAGE		
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

SSS__XXSSCFB08	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.5.1**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB	
Configuration:	The **user B** has subscribed to CFB and CDIVN	
Selection criteria:	CFB and CDIVN supported.	
Test purpose:	Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B, having activated the CDIVN service, receives a NOTIFY request indicating the call diversion.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	
Comments:		

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
		Start Activation CDIVN				
			←	SUBSCRIBE		
			→	200 OK SUBSCRIBE		
			→	NOTIFY		
			←	200 OK NOTIFY		
		End Activation CDIVN				
INVITE	→		→	INVITE		
			←	486 Busy Here		
			→	ACK		
		Communication diversion is performed				
181 Call Is Being Forwarded (optional)	←				→	INVITE
			→	NOTIFY		
			←	200 OK NOTIFY		
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

SSS__XXSSCFB09	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB	
Configuration:	The **user B** has subscribed to CFB **Subscription options:** Originating user receives notification that his communication has been diverted = **No**	
Selection criteria:	CFB supported.	
Test purpose:	Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded message.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	
Comments:		

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
			←	486 Busy Here		
			→	ACK		
		Communication diversion is performed				
					→	INVITE
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

SSS__XXSSCFB10	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB
Configuration:	The **user B** has subscribed to CFB and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **No** Served user allows the presentation of his/her URI to originating user in diversion notification = **No**
Selection criteria:	CFB supported.
Test purpose:	Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a Privacy header with value "id" and not containing a P-Asserted-Identity indicating the URI of user B and not containing a History-Info header (with CDIV related cause value) indicating the URI of user B or user A.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
			←	486 Busy Here		
			→	ACK		
		Communication diversion is performed				
181 Call Is Being Forwarded	←				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

SSS__XXSSCFB11	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB	
Configuration:	The **user B** has subscribed to CFB and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **Yes** Served user allows the presentation of his/her URI to originating user in diversion notification = **Yes**	
Selection criteria:	CFB supported.	
Test purpose:	Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a P-Asserted-Identity indicating the URI of user B and containing a History-Info header including a first entry with the hi-targeted-to-URI of user B, index = 1, cause param = 486 and including a second entry with the hi-targeted-to-URI of user C, index = 1.1	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
			←	486 Busy Here		
			→	ACK		
		Communication diversion is performed				
181 Call Is Being Forwarded	←				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

SSS__XXSSCFB12	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.2.2**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB	
Configuration:	The **user B** has subscribed to CFB and has not activated OIR **Subscription options:** Served user allows the presentation of his/her URI to diverted-to user = **Yes**	
Selection criteria:	CFB supported.	
Test purpose:	Ensure that when user A calls user B which is user determined user busy (UDUB), the call is forwarded to user C. Ensure that User C receives an INVITE message containing a History-Info header (with CDIV related cause value) including an entry with the hi-targeted-to-URI of user B.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
			←	486 Busy Here		
			→	ACK		
		Communication diversion is performed				
181 Call Is Being Forwarded	←				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

6.2.6.3 CFNR

SSS__XXSSCFNR01	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.5**			
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNR			
Configuration:	The **user B** has subscribed to CFNR, CDIVN is not activated **Subscription options:** Served user receives indication that a communication has been forwarded = **Yes**			
Selection criteria:	CFNR supported.			
Test purpose:	Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B receives a MESSAGE request indicating the call diversion.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT			
Comments:				
SIP UA A	**SUT**		**SIP UA B**	**SIP UA C**
INVITE	→	→	INVITE	
180 Ringing	←	←	180 Ringing	
No reply timer expires - Communication diversion is performed				
181 Call Is Being Forwarded (optional)	←	→	CANCEL (Note)	→ INVITE
		←	200 OK CANCEL	
		←	487 Request Terminated	
		→	ACK	
		→	MESSAGE	
		←	200 OK MESSAGE	
				← 180 Ringing
200 OK INVITE	←			← 200 OK INVITE
ACK	→			→ ACK
Communication				
BYE	→			→ BYE
200 OK BYE	←			← 200 OK BYE
NOTE: The communication to user B may be retained until the 180 Ringing from user C has been received.				

SSS__XXSSCFNR02	**CDIV reference to:** **TS 124 504 [12], clause 4.5.2.6.5.1**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNR	
Configuration:	The **user B** has subscribed to CFNR and CDIVN	
Selection criteria:	CFNR and CDIVN supported.	
Test purpose:	Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B, having activated the CDIVN service, receives a NOTIFY request indicating the call diversion.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	

Comments:

SIP UA A		**SUT**	**SIP UA B**		**SIP UA C**
		Start Activation CDIVN			
		←	SUBSCRIBE		
		→	200 OK SUBSCRIBE		
		→	NOTIFY		
		←	200 OK NOTIFY		
		End Activation CDIVN			
INVITE	→	→	INVITE		
180 Ringing	←	←	180 Ringing		
		No reply timer expires - Communication diversion is performed			
181 Call Is Being Forwarded (optional)	←	→	CANCEL (Note)	→	INVITE
		←	200 OK CANCEL		
		←	487 Request Terminated		
		→	ACK		
		→	NOTIFY		
		←	200 OK NOTIFY		
				←	180 Ringing
200 OK INVITE	←			←	200 OK INVITE
ACK	→			→	ACK
		Communication			
BYE	→			→	BYE
200 OK BYE	←			←	200 OK BYE

NOTE: The communication to user B may be retained until the 180 Ringing from user C has been received.

SSS__XXSSCFNR03	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNR
Configuration:	The **user B** has subscribed to CFNR **Subscription options:** Originating user receives notification that his communication has been diverted = **No**
Selection criteria:	CFNR supported.
Test purpose:	Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded message.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
No reply timer expires - Communication diversion is performed						
			→	CANCEL (Note)	→	INVITE
			←	200 OK CANCEL		
			←	487 Request Terminated		
			→	ACK		
					←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

NOTE: The communication to user B may be retained until the 180 Ringing from user C has been received.

SSS__XXSSCFNR04	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNR
Configuration:	The **user B** has subscribed to CFNR and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **No** Served user allows the presentation of his/her URI to originating user in diversion notification = **No**
Selection criteria:	CFNR supported.
Test purpose:	Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a Privacy header with value "id" and not containing a P-Asserted-Identity indicating the URI of user B and not containing a History-Info header indicating the URI of user B or user A.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
No reply timer expires - Communication diversion is performed						
181 Call Is Being Forwarded	←		→	CANCEL (Note)	→	INVITE
			←	200 OK CANCEL		
			←	487 Request Terminated		
			→	ACK		
					←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

NOTE: The communication to user B may be retained until the 180 Ringing from user C has been received.

SSS__XXSSCFNR05	**CDIV reference to:** **TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNR
Configuration:	The **user B** has subscribed to CFNR and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **Yes** Served user allows the presentation of his/her URI to originating user in diversion notification = **Yes**
Selection criteria:	CFNR supported.
Test purpose:	Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a P-Asserted-Identity indicating the URI of user B and containing a History-Info header including a first entry with the hi-targeted-to-URI of user B, index = 1, cause param = 408 and including a second entry with the hi-targeted-to-URI of user C, index = 1.1
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
No reply timer expires - Communication diversion is performed						
181 Call Is Being Forwarded	←		→	CANCEL (Note)	→	INVITE
			←	200 OK CANCEL		
			←	487 Request Terminated		
			→	ACK		
					←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

NOTE: The communication to user B may be retained until the 180 Ringing from user C has been received.

SSS__XXSSCFNR06	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.2.2**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNR
Configuration:	The **user B** has subscribed to CFNR and has not activated OIR **Subscription options:** Served user allows the presentation of his/her URI to diverted-to user = **Yes**
Selection criteria:	CFNR supported.
Test purpose:	Ensure that when user A calls user B which does not answer, the call is forwarded to user C. Ensure that User C receives an INVITE message containing a History-Info header (with CDIV related cause value) including an entry with the hi-targeted-to-URI of user B.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
No reply timer expires - Communication diversion is performed						
181 Call Is Being Forwarded	←		→	CANCEL (Note)	→	INVITE
			←	200 OK CANCEL		
			←	487 Request Terminated		
			→	ACK		
					←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

NOTE: The communication to user B may be retained until the 180 Ringing from user C has been received.

SSS__XXSSCFNR07	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.3 3)**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNR
Configuration:	The **user B** has subscribed to CFNR **Subscription options:** Served user communication retention on invocation of diversion = **Retain communication to the served user until alerting begins at the diverted-to user**
Selection criteria:	CFNR supported.
Test purpose:	Ensure that when user A calls user B which has not answered before the expiry of the No reply timer, and when the communication has been forwarded to user C and when user B answers the communication before user C starts alerting, the communication is established between user A and user B and the communication is cancelled towards user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters).
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
No reply timer expires - Communication diversion is performed						
181 Call Is Being Forwarded (optional)	←				→	INVITE
200 OK INVITE	←		←	200 OK INVITE		
ACK	→		→	ACK	→	CANCEL
					←	200 OK CANCEL
			Communication			
BYE	→		→	BYE		
200 OK BYE	←		←	200 OK BYE		

6.2.6.4 CFNRc

SSS__XXSSCFNRc01	**CDIV reference to:** **TS 124 504 [12], clause 4.5.2.6.5.1**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNRc	
Configuration:	The **user B** has subscribed to CFNRc and CDIVN	
Selection criteria:	CFNRC and CDIVN supported	
Test purpose:	Ensure that when user A calls user B which is unreachable, the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B, having activated the CDIVN service, receives a NOTIFY request indicating the call diversion.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	
Comments:		

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
		Activation CDIVN				
			←	SUBSCRIBE		
			→	200 OK SUBSCRIBE		
			→	NOTIFY		
			←	200 OK NOTIFY		
		User B becomes "Not reachable" (indication not specified)				
INVITE	→					
		Communication diversion is performed				
181 Call Is Being Forwarded (optional)	←				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE
		User B becomes "Reachable"				
			→	NOTIFY		
			←	200 OK NOTIFY		

SSS__XXSSCFNRc02	**CDIV reference to:** **TS 124 504 [12], clause 4.5.2.6.4**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNRc	
Configuration:	The **user B** has subscribed to CFNRc **Subscription options:** Originating user receives notification that his communication has been diverted = **No**	
Selection criteria:	CFNRC supported	
Test purpose:	Ensure that when user A calls user B which is unreachable, the call is forwarded to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded message.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	
Comments:		

SIP UA A		**SUT**		**SIP UA C**
INVITE	→		→	INVITE
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SSS__XXSSCFNRc03	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNRc
Configuration:	The **user B** has subscribed to CFNRc and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **No** Served user allows the presentation of his/her URI to originating user in diversion notification = **No**
Selection criteria:	CFNRC supported
Test purpose:	Ensure that when user A calls user B which is unreachable, the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a Privacy header with value "id" and not containing a P-Asserted-Identity indicating the URI of user B and not containing a History-Info header (with CDIV related cause value) indicating the URI of user B or user A.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT
Comments:	

SIP UA A		**SUT**		**SIP UA C**
INVITE	→		→	INVITE
181 Call Is Being Forwarded	←			
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SSS__XXSSCFNRc04	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNRc
Configuration:	The **user B** has subscribed to CFNRc and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **Yes** Served user allows the presentation of his/her URI to originating user in diversion notification = **Yes**
Selection criteria:	CFNRC supported
Test purpose:	Ensure that when user A calls user B which is unreachable, the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a P-Asserted-Identity indicating the URI of user B and containing a History-Info header including a first entry with the hi-targeted-to-URI of user B, index = 1, cause param = 503 and including a second entry with the hi-targeted-to-URI of user C, index = 1.1
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT
Comments:	

SIP UA A		**SUT**		**SIP UA C**
INVITE	→		→	INVITE
181 Call Is Being Forwarded	←			
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SSS__XXSSCFNRc05	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.2.2**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNRc
Configuration:	The **user B** has subscribed to CFNRc and has not activated OIR **Subscription options:** Served user allows the presentation of his/her URI to diverted-to user = **Yes**
Selection criteria:	CFNRC supported
Test purpose:	Ensure that when user A calls user B which is unreachable, the call is forwarded to user C. Ensure that User C receives an INVITE message containing a History-Info header including an entry (with CDIV related cause value) with the hi-targeted-to-URI of user B.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT
Comments:	

SIP UA A		**SUT**		**SIP UA C**
INVITE	→		→	INVITE
181 Call Is Being Forwarded (optional)	←			
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

6.2.6.5 CFNL

SSS__XXSSCFNL01	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.5.1**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNL
Configuration:	The **user B** has subscribed to CFNL and CDIVN
Selection criteria:	CFNL and CDIVN supported
Test purpose:	Ensure that when user A calls user B which is not logged in, the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B, having activated the CDIVN service, receives a NOTIFY request indicating the call diversion.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT
Comments:	

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
		Activation CDIVN				
			←	SUBSCRIBE		
			→	200 OK SUBSCRIBE		
			→	NOTIFY		
			←	200 OK NOTIFY		
		Log off User B				
INVITE	→					
		Communication diversion is performed				
181 Call Is Being Forwarded (optional)	←				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE
		Log in User B				
			→	NOTIFY		
			←	200 OK NOTIFY		

SSS__XXSSCFNL02	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNL	
Configuration:	The **user B** has subscribed to CFNL **Subscription options:** Originating user receives notification that his communication has been diverted = **No**	
Selection criteria:	CFNL supported	
Test purpose:	Ensure that when user A calls user B which is not logged in, the call is forwarded to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded message.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	
Comments:		

SIP UA A		**SUT**		**SIP UA C**
INVITE	→		→	INVITE
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SSS__XXSSCFNL03	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNL	
Configuration:	The **user B** has subscribed to CFNL and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **No** Served user allows the presentation of his/her URI to originating user in diversion notification = **No**	
Selection criteria:	CFNL supported	
Test purpose:	Ensure that when user A calls user B which is not logged in, the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a Privacy header with value "id" and not containing a P-Asserted-Identity indicating the URI of user B and not containing a History-Info header (with CDIV related cause value) indicating the URI of user B or user A.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	
Comments:		

SIP UA A		**SUT**		**SIP UA C**
INVITE	→		→	INVITE
181 Call Is Being Forwarded	←			
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SSS__XXSSCFNL04	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNL
Configuration:	The **user B** has subscribed to CFNL and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **Yes** Served user allows the presentation of his/her URI to originating user in diversion notification = **Yes**
Selection criteria:	CFNL supported
Test purpose:	Ensure that when user A calls user B which is not logged in, the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a P-Asserted-Identity indicating the URI of user B and containing a History-Info header including a first entry with the hi-targeted-to-URI of user B, index = 1, cause param = 404 and including a second entry with the hi-targeted-to-URI of user C, index = 1.1
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT
Comments:	

SIP UA A		**SUT**		**SIP UA C**
INVITE	→		→	INVITE
181 Call Is Being Forwarded	←			
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SSS__XXSSCFNL05	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.2.2**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFNL
Configuration:	The **user B** has subscribed to CFNL and has not activated OIR **Subscription options:** Served user allows the presentation of his/her URI to diverted-to user = **Yes**
Selection criteria:	CFNL supported
Test purpose:	Ensure that when user A calls user B which is not logged in, the call is forwarded to user C. Ensure that User C receives an INVITE message containing a History-Info header (with CDIV related cause value) including an entry with the hi-targeted-to-URI of user B.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT
Comments:	

SIP UA A		**SUT**		**SIP UA C**
INVITE	→		→	INVITE
181 Call Is Being Forwarded (optional)	←			
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

6.2.6.6 CD

6.2.6.6.1 CD Immediate

SSS__XXSSCD01	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.5**			
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB			
Configuration:	The **user B** has subscribed to CD, CDIVN is not activated **Subscription options:** Served user receives indication that a communication has been forwarded = **Yes**			
Selection criteria:	CD supported.			
Test purpose:	Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B receives a MESSAGE request indicating the call diversion.			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT			
Comments:				
SIP UA A		**SUT**	**SIP UA B**	**SIP UA C**
INVITE	→		→ INVITE	
			← 302 Moved Temporarily	
			→ ACK	
		Communication diversion is performed		
181 Call Is Being Forwarded (optional)	←			→ INVITE
			→ MESSAGE	
			← 200 OK MESSAGE	
180 Ringing	←			← 180 Ringing
200 OK INVITE	←			← 200 OK INVITE
ACK	→			→ ACK
		Communication		
BYE	→			→ BYE
200 OK BYE	←			← 200 OK BYE

SSS__XXSSCD02	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.5.1**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CFB
Configuration:	The **user B** has subscribed to CD and CDIVN
Selection criteria:	CD and CDIVN supported.
Test purpose:	Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters). Ensure that User B, having activated the CDIVN service, receives a NOTIFY request indicating the call diversion.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
		Start Activation CDIVN				
			←	SUBSCRIBE		
			→	200 OK SUBSCRIBE		
			→	NOTIFY		
			←	200 OK NOTIFY		
		End Activation CDIVN				
INVITE	→		→	INVITE		
			←	302 Moved Temporarily		
			→	ACK		
		Communication diversion is performed				
181 Call Is Being Forwarded (optional)	←				→	INVITE
			→	NOTIFY		
			←	200 OK NOTIFY		
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

SSS__XXSSCD03	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CD
Configuration:	The **user B** has subscribed to CD **Subscription options:** Originating user receives notification that his communication has been diverted = **No**
Selection criteria:	CD supported.
Test purpose:	Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded message.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
			←	302 Moved Temporarily		
			→	ACK		
		Communication deflection is performed				
					→	INVITE
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

SSS__XXSSCD04	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**		
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CD		
Configuration:	The **user B** has subscribed to CD and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **No** Served user allows the presentation of his/her URI to originating user in diversion notification = **No**		
Selection criteria:	CD supported.		
Test purpose:	Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a Privacy header with value "id" and not containing a P-Asserted-Identity indicating the URI of user B and not containing a History-Info header indicating the URI of user B or user A.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT		
Comments:			
SIP UA A	**SUT**	**SIP UA B**	**SIP UA C**
INVITE	→	→ INVITE	
		← 302 Moved Temporarily	
		→ ACK	
	Communication deflection is performed		
181 Call Is Being Forwarded	←		→ INVITE
180 Ringing	←		← 180 Ringing
200 OK INVITE	←		← 200 OK INVITE
ACK	→		→ ACK
BYE	→		→ BYE
200 OK BYE	←		← 200 OK BYE

SSS__XXSSCD05	CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CD
Configuration:	The **user B** has subscribed to CD and has not activated TIR **Subscription options:** Originating user receives notification that his communication has been diverted = **Yes** Served user allows the presentation of diverted to URI to originating user in diversion notification = **Yes** Served user allows the presentation of his/her URI to originating user in diversion notification = **Yes**
Selection criteria:	CD supported.
Test purpose:	Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message containing a P-Asserted-Identity indicating the URI of user B and containing a History-Info header including a first entry with the hi-targeted-to-URI of user B, index = 1, cause param = 480 and including a second entry with the hi-targeted-to-URI of user C, index = 1.1 Note: "index of these new H-I entries may be different if other entries have been added to H-I header."
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
			←	302 Moved Temporarily		
			→	ACK		
		Communication deflection is performed				
181 Call Is Being Forwarded	←				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

SSS__XXSSCD06	CDIV reference to: TS 124 504 [12], clause 4.5.2.6.2.2
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CD
Configuration:	The **user B** has subscribed to CD and has not activated OIR **Subscription options:** Served user allows the presentation of his/her URI to diverted-to user = **Yes**
Selection criteria:	CD supported.
Test purpose:	Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C. Ensure that User C receives an INVITE message containing a History-Info header (with CDIV related cause value) including an entry with the hi-targeted-to-URI of user B.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
			←	302 Moved Temporarily		
			→	ACK		
		Communication deflection is performed				
181 Call Is Being Forwarded	←				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

6.2.6.6.2 CD during alerting

SSS__XXSSCD07	**CDIV reference to: TS 124 504 [12], clause 4.5.2.6.4**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CD	
Configuration:	The **user B** has subscribed to CD **Subscription options:** Originating user receives notification that his communication has been diverted = **No**	
Selection criteria:	CD supported.	
Test purpose:	Ensure that when user A calls user B which deflects the communication towards user C during alerting, the call is forwarded to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded message.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT	
Comments:		

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
			←	302 Moved Temporarily		
			→	ACK		
		Communication deflection is performed				
					→	INVITE
					←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

6.2.7 Test purposes for CONF

6.2.7.1 Conference creation

SSS__XXSSCONF_CRE_001	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_CRE	
Configuration:	CONF	
Selection criteria:	Conference creation by Three-way session creation. REFER request to the user, Conference event package is subscribed.	

<table>
<tr><th>SSS__XXSSCONF_CRE_001</th><th>CONF reference to:
TS 124 147 [19], clauses 5.2.1, 5.3.1.3</th><th></th></tr>
<tr><td>Test purpose:</td><td colspan="2">Creation of the conference
Ensure that, when User A sends an INVITE request with request URI set to a valid conference factory URI:
• User A receives a 200 OK SIP response from the conference focus containing "isfocus" feature parameter in Contact header. User A shall store the content of the receive Contact header as the conference URI.
• User A sends an ACK SIP request.
• User A sends a SUBSCRIBE request with request URI set to the conference URI (previously stored) and the Event header set to "conference".
• User A receives a 200 OK SIP response to the SUBSCRIBE request.
• User A receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference".
• User A sends a 200 OK SIP response to the NOTIFY request.

Inviting users to the conference
For each active SIP session, User A sends a REFER request to the remote user (User B or User C) with request URI set to the URI of the address of the remote user and Refer-To header set to the conference URI previously stored (the parameter "method" set to INVITE in the Refer-To header can be included or omitted):
• Remote user receives a REFER request containing the Refer-To header set to the conference URI.
• Remote user sends a 202 Accepted SIP response to the REFER request.
• User A receives a 202 Accepted SIP response to the REFER request.
• Remote user sends an INVITE request with request URI set to conference URI to the conference focus.
• Remote user sends a NOTIFY request to the User A (on the same dialog of the REFER previously received) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 100 Trying.
• User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 100 Trying.
• User A sends a 200 OK SIP response to the NOTIFY request.
• Remote user receives a 200 OK SIP response to the NOTIFY request.
• Remote user receives a 200 OK SIP response to the INVITE request from the conference focus.
• Remote user sends an ACK to the conference focus.
• Remote user sends a NOTIFY request to the User A (on the same dialog of the REFER previously received) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 200 OK.
• User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 200 OK.
• User A sends a 200 OK SIP response to the NOTIFY request.
• Remote user receives a 200 OK SIP response to the NOTIFY request.
• User A sends a BYE request to the remote user in order to release the active SIP session between the user A and the remote user.
• Remote user receives a BYE request from user A.
• Remote user sends a 200 OK SIP response to the BYE request.
• User A receives a 200 OK SIP response to the BYE request.
• User A receives a NOTIFY from the conference focus (on the same dialog of the SUBSCRIBE previously sent).
• User A sends a 200 OK SIP response to the NOTIFY request.

NOTE: Additionally, User A may include the Referred-By header to the REFER and set it to his SIP URI.</td></tr>
</table>

SSS__XXSSCONF_CRE_001	**CONF reference to:** **TS 124 147 [19], clauses 5.2.1, 5.3.1.3**	
Precondition:	• User A was participating in two SIP sessions (one with User B and the other with User C). • The SIP session between User A and User B was previously put on HOLD by User A.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** INVITE: Request URI contains the *conference factory URI* 200 OK: "isfocus" feature parameter indicated in Contact header field *conference URI* contains in the Contact header field SUBSCRIBE: Request URI contains the *conference URI*, Event header contains "conference" REFER: Refer-to header contains the conference URI NOTIFY : Event header contains **conference**; Subscription-State header contains **active,** application/conference-info+xml contains connected, dialled-in NOTIFY 1: Event header contains **refer**; Subscription-State header contains **active,** Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying** NOTIFY 2: Event header contains **refer**; Subscription-State header contains **terminated,** Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK** NOTIFY 3: Event header contains **conference**; Subscription-State header contains **active,** application/conference-info+xml contains connected, dialled-in NOTIFY 4: Event header contains **refer**; Subscription-State header contains **active,** Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying** NOTIFY 5: Event header contains **refer**; Subscription-State header contains **terminated,** Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK** NOTIFY 6: Event header contains **conference**; Subscription-State header contains **active,** application/conference-info+xml contains connected, dialled-in	

Comments:

SIP UA A		**Focus**		**SIP UA B**		**SIP UA C**
		Establishment of session #1				
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
200 OK (INVITE)	←		←	200 OK (INVITE)		
ACK	→		→	ACK		
INVITE (sendonly)	→		→	INVITE (sendonly)		
200 OK (recvonly)	←		←	200 OK (recvonly)		
ACK	→		→	ACK		
		Establishment of session #2				
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	←				←	200 OK (INVITE)
ACK	→				→	ACK
		Conference creation				
INVITE	→	INVITE				
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→	ACK				
SUBSCRIBE	→	SUBSCRIBE				
200 OK (SUBSCRIBE)	←	200 OK (SUBSCRIBE)				
NOTIFY	←	NOTIFY				
200 OK (NOTIFY)	→	200 OK (NOTIFY)				

SSS__XXSSCONF_CRE_001	CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3					
		Inviting UA B to the conference				
REFER	→		→	REFER		
202 Accepted	←		←	202 Accepted		
		INVITE	←	INVITE		
NOTIFY 1	←		←	NOTIFY 1		
200 OK (NOTIFY 1)	→		→	200 OK (NOTIFY 1)		
		200 OK	→	200 OK		
		ACK	←	ACK		
NOTIFY 2	←		←	NOTIFY 2		
200 OK (NOTIFY 2)	→		→	200 OK (NOTIFY 2)		
BYE	→		→	BYE		
200 OK (BYE)	←		←	200 OK (BYE)		
NOTIFY 3	←	NOTIFY 3				
200 OK (NOTIFY 3)	→	200 OK (NOTIFY 3)				
		Inviting UA C to the conference				
REFER	→				→	REFER
202 Accepted	←				←	202 Accepted
		INVITE	←		←	INVITE
NOTIFY 4	←				←	NOTIFY 4
200 OK (NOTIFY 4)	→				→	200 OK (NOTIFY 4)
		200 OK	→		→	200 OK
		ACK	←		←	ACK
NOTIFY 5	←				←	NOTIFY 5
200 OK (NOTIFY 5)	→				→	200 OK (NOTIFY 5)
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)
NOTIFY 6	←	NOTIFY 6				
200 OK (NOTIFY 6)	→	200 OK (NOTIFY 6)				

<table>
<tr><th>SSS__XXSSCONF
_CRE_002</th><th>CONF reference to:
TS 124 147 [19], clauses 5.2.1, 5.3.1.3</th><th></th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-SIP-SIP/Supplementary_Services/CONF_CRE</td></tr>
<tr><td>Configuration:</td><td colspan="2">CONF</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">Conference creation by Three-way session creation. REFER request to the user, Conference event package not subscribed.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Creation of the conference
Ensure that, when User A sends an INVITE request with request URI set to a valid conference factory URI:
• User A receives a 200 OK SIP response from the conference focus containing "isfocus" feature parameter in Contact header. User A shall store the content of the receive Contact header as the conference URI.
• User A sends an ACK SIP request.

Inviting users to the conference
For each active SIP session, User A sends a REFER request to the remote user (User B or User C) with request URI set to the URI of the address of the remote user and Refer-To header set to the conference URI previously stored (the parameter "method" set to INVITE in the Refer-To header can be included or omitted):
• Remote user receives a REFER request containing the Refer-To header set to the conference URI.
• Remote user sends a 202 Accepted SIP response to the REFER request.
• User A receives a 202 Accepted SIP response to the REFER request.
• Remote user sends an INVITE request with request URI set to conference URI to the conference focus.
• Remote user sends a NOTIFY request to the User A (on the same dialog of the REFER previously received) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 100 Trying.
• User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 100 Trying.
• User A sends a 200 OK SIP response to the NOTIFY request.
• Remote user receives a 200 OK SIP response to the NOTIFY request.
• Remote user receives a 200 OK SIP response to the INVITE request from the conference focus.
• Remote user sends an ACK to the conference focus.
• Remote user sends a NOTIFY request to the User A (on the same dialog of the REFER previously received) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 200 OK.
• User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 200 OK.
• User A sends a 200 OK SIP response to the NOTIFY request.
• Remote user receives a 200 OK SIP response to the NOTIFY request.
• User A sends a BYE request to the remote user in order to release the active SIP session between the user A and the remote user.
• Remote user receives a BYE request from user A.
• Remote user sends a 200 OK SIP response to the BYE request.
• User A receives a 200 OK SIP response to the BYE request.

NOTE: Additionally, User A may include the Referred-By header to the REFER and set it to his SIP URI.</td></tr>
<tr><td>Precondition:</td><td colspan="2">• User A was participating in two SIP sessions (one with User B and the other with User C).
• The SIP session between User A and User B was previously put on HOLD by User A.</td></tr>
</table>

SSS__XXSSCONF _CRE_002	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3**	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** INVITE: Request URI contains the *conference factory URI*. 200 OK: "isfocus" feature parameter indicated in Contact header field conference URI contains in the Contact header field. REFER: Refer-to header contains the *conference URI*. NOTIFY 1: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains SIP/2.0 100 Trying. NOTIFY 2: Event header contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK**. NOTIFY 3: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying**. NOTIFY 4: Event header contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK**.	

Comments:

SIP UA A		**Focus**		**SIP UA B**		**SIP UA C**
		Establishment of session #1				
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
200 OK (INVITE)	←		←	200 OK (INVITE)		
ACK	→		→	ACK		
INVITE (sendonly)	→		→	INVITE (sendonly)		
200 OK (recvonly)	←		←	200 OK (recvonly)		
ACK	→		→	ACK		
		Establishment of session #2				
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	←				←	200 OK (INVITE)
ACK	→				→	ACK
		Conference creation				
INVITE	→	INVITE				
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→	ACK				
		Inviting UA B to the conference				
REFER	→		→	REFER		
202 Accepted	←		←	202 Accepted		
		INVITE	←	INVITE		
NOTIFY 1	←		←	NOTIFY 1		
200 OK (NOTIFY 1)	→		→	200 OK (NOTIFY 1)		
		200 OK	→	200 OK (INVITE)		
		ACK	←	ACK		
NOTIFY 2	←		←	NOTIFY 2		
200 OK (NOTIFY 2)	→		→	200 OK (NOTIFY 2)		
BYE	→		→	BYE		
200 OK (BYE)	←		←	200 OK (BYE)		
		Inviting UA C to the conference				
REFER	→				→	REFER
202 Accepted	←				←	202 Accepted
		INVITE	←		←	INVITE
NOTIFY 3	←				←	NOTIFY 3
200 OK (NOTIFY 3)	→				→	200 OK (NOTIFY 3)
		200 OK	→		→	200 OK
		ACK	←		←	ACK
NOTIFY 4	←				←	NOTIFY 4
200 OK (NOTIFY 4)	→				→	200 OK (NOTIFY 4)
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)

SSS__XXSSCONF_ CRE_003	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_CRE	
Configuration:	CONF	
Selection criteria:	Conference creation by Three-way session creation. REFER request to the conference focus, Conference event package subscribed.	
Test purpose:	**Creation of the conference** Ensure that, when User A sends an INVITE request with request URI set to a valid *conference factory URI*: • User A receives a 200 OK SIP response from the *conference focus* containing "isfocus" feature parameter in Contact header. User A shall store the content of the receive Contact header as the *conference URI*. • User A sends an ACK SIP request. • User A sends a SUBSCRIBE request with request URI set to the *conference URI* (previously stored) and the Event header set to "conference". • User A receives a 200 OK SIP response to the SUBSCRIBE request. • User A receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference". • User A sends a 200 OK SIP response to the NOTIFY request. **Inviting users to the conference** For each active SIP session, User A sends a REFER request to the conference focus with request URI set to the *conference URI* previously stored and Refer-To header set to the SIP URI of the remote user (the parameter "method" set to INVITE in the Refer-To header can be included or omitted): • User A receives a 202 Accepted SIP response to the REFER request. • Remote user receives an INVITE request from the conference focus to be invited to the conference. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A sends a 200 OK SIP response to the NOTIFY request. • Remote user sends a 200 OK SIP response to the INVITE request from the conference focus. • Remote user receives an ACK from the conference focus. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK**. • User A sends a 200 OK SIP response to the NOTIFY request. • User A sends a BYE request to the remote user in order to release the active SIP session between the user A and the remote user. • Remote user receives a BYE request from user A. • Remote user sends a 200 OK SIP response to the BYE request. • User A receives a 200 OK SIP response to the BYE request. • User A receives a NOTIFY from the conference focus (on the same dialog of the SUBSCRIBE previously sent). • User A sends a 200 OK SIP response to the NOTIFY request. NOTE: Additionally, User A may include the Referred-By header to the REFER and set it to his SIP URI.	
Precondition:	• User A was participating in two SIP sessions (one with User B and the other with User C). • The SIP session between User A and User B was previously put on HOLD by User A.	

SSS__XXSSCONF_ CRE_003	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3**	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** INVITE: Request URI contains the *conference factory URI*. 200 OK: "isfocus" feature parameter indicated in Contact header field conference URI contains in the Contact header field. SUBSCRIBE: Request URI contains the *conference URI*. REFER 1: Refer-to header contains the SIP URI of the **UA B.** NOTIFY: Event header contains conference; Subscription-State header contains active, application/conference-info+xml contains connected, dialled-in NOTIFY 1: Event header contains **refer**; Subscription-State header contains **active,** Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 2: Event header contains **refer**; Subscription-State header contains **terminated,** Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.** NOTIFY 3: Event header contains **conference**; Subscription-State header contains **active,** application/conference-info+xml contains connected, dialled-out. REFER 2: Refer-to header contains the URI of **the UA C.** NOTIFY 4: Event header contains **refer**; Subscription-State header contains **active,** Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 5: Event header contains **refer**; Subscription-State header contains **terminated,** Content-Type header contains "message/sipfrag", message/sipfrag contains **SIP/2.0 200 OK.** NOTIFY 6: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml contains connected, dialled-out.	

Comments:

SIP UA A		**Focus**		**SIP UA B**		**SIP UA C**
		Establishment of session #1				
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
200 OK (INVITE)	←		←	200 OK (INVITE)		
ACK	→		→	ACK		
INVITE (sendonly)	→		→	INVITE(sendonly)		
200 OK (recvonly)	←		←	200 OK (recvonly)		
ACK	→		→	ACK		
		Establishment of session #2				
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	←				←	200 OK (INVITE)
ACK	→				→	ACK
		Conference creation				
INVITE	→	INVITE				
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→	ACK				
SUBSCRIBE	→	SUBSCRIBE				
200 OK (SUBSCRIBE)	←	200 OK (SUBSCRIBE)				
NOTIFY	←	NOTIFY				
200 OK (NOTIFY)	→	200 OK (NOTIFY)				
		Inviting UA B to the conference				
REFER 1	→	REFER 1				
202 Accepted	←	202 Accepted				
		INVITE	→	INVITE		
NOTIFY 1	←	NOTIFY 1				
200 OK (NOTIFY 1)	→	200 OK (NOTIFY 1)				
		200 OK (INVITE)	←	200 OK (INVITE)		
		ACK	→	ACK		
NOTIFY 2	←	NOTIFY 2				
200 OK (NOTIFY 2)	→	200 OK (NOTIFY 2)				
BYE	→		→	BYE		
200 OK (BYE)	←		←	200 OK (BYE)		
NOTIFY 3	←	NOTIFY 3				
200 OK (NOTIFY 3)	→	200 OK (NOTIFY 3)				

SSS__XXSSCONF_ CRE_003		CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3		
		Inviting UA C to the conference		
REFER 2	→	REFER 2		
202 Accepted	←	202 Accepted		
		INVITE →	→	INVITE
NOTIFY 4	←	NOTIFY 4		
200 OK (NOTIFY 4)	→	200 OK (NOTIFY 4)		
		200 OK (INVITE) ←	←	200 OK (INVITE)
		ACK →	→	ACK
NOTIFY 5	←	NOTIFY 5		
200 OK (NOTIFY 5)	→	200 OK (NOTIFY 5)		
BYE	→		→	BYE
200 OK (BYE)	←		←	200 OK (BYE)
NOTIFY 6	←	NOTIFY 6		
200 OK (NOTIFY 6)	→	200 OK (NOTIFY 6)		

SSS__XXSSCONF_ CRE_004	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_CRE	
Configuration:	CONF	
Selection criteria:	Conference creation by Three-way session creation. REFER request to the focus, Conference event package not subscribed.	
Test purpose:	**Creation of the conference** Ensure that, when User A sends an INVITE request with request URI set to a valid *conference factory URI*: • User A receives a 200 OK SIP response from the *conference focus* containing "isfocus" feature parameter in Contact header. User A shall store the content of the receive Contact header as the *conference URI*. • User A sends an ACK SIP request. **Inviting users to the conference** For each active SIP session, User A sends a REFER request to the conference focus with request URI set to the *conference URI* previously stored and Refer-To header set to the SIP URI of the remote user (the parameter "method" set to INVITE in the Refer-To header can be included or omitted): • User A receives a 202 Accepted SIP response to the REFER request. • Remote user receives an INVITE request from the conference focus to be invited to the conference. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A sends a 200 OK SIP response to the NOTIFY request. • Remote user sends a 200 OK SIP response to the INVITE request from the conference focus. • Remote user receives an ACK from the conference focus. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK**. • User A sends a 200 OK SIP response to the NOTIFY request. • User A sends a BYE request to the remote user in order to release the active SIP session between the user A and the remote user. • Remote user receives a BYE request from user A. • Remote user sends a 200 OK SIP response to the BYE request. • User A receives a 200 OK SIP response to the BYE request. NOTE: Additionally, User A may include the Referred-By header to the REFER and set it to his SIP URI.	

SSS__XXSSCONF_ CRE_004	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3**	
Precondition:	• User A was participating in two SIP sessions (one with User B and the other with User C). • The SIP session between User A and User B was previously put on HOLD by User A.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** INVITE: Request URI contains the *conference factory URI*. 200 OK: "isfocus" feature parameter indicated in Contact header field conference URI contains in the Contact header field. REFER 1: Refer-to header contains the SIP URI of the **UA B.** NOTIFY 1: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 2: Event header contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.** REFER 2: Refer-to header contains the URI of the **UA C.** NOTIFY 3: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 4: Event header contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.**	

Comments:

SIP UA A		**Focus**		**SIP UA B**		**SIP UA C**
		Establishment of session #1				
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
200 OK (INVITE)	←		←	200 OK (INVITE)		
ACK	→		→	ACK		
INVITE (sendonly)	→		→	INVITE (sendonly)		
200 OK (recvonly)	←		←	200 OK (recvonly)		
ACK	→		→	ACK		
		Establishment of session #2				
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	←				←	200 OK (INVITE)
ACK	→				→	ACK
		Conference creation				
INVITE	→	INVITE				
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→	ACK				
		Inviting UA B to the conference				
REFER 1	→	REFER 1				
202 Accepted	←	202 Accepted				
		INVITE	→	INVITE		
NOTIFY 1	←	NOTIFY 1				
200 OK (NOTIFY 1)	→	200 OK (NOTIFY 1)				
		200 OK (INVITE)	←	200 OK (INVITE)		
		ACK	→	ACK		
NOTIFY 2	←	NOTIFY 2				
200 OK (NOTIFY 2)	→	200 OK (NOTIFY 2)				
BYE	→		→	BYE		
200 OK (BYE)	←		←	200 OK (BYE)		

SSS__XXSSCONF_ CRE_004	CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3			
		Inviting UA C to the conference		
REFER 2	→	REFER 2		
202 Accepted	←	202 Accepted		
		INVITE →	→	INVITE
NOTIFY 3	←	NOTIFY 3		
200 OK (NOTIFY 3)	→	200 OK (NOTIFY 3)		
		200 OK (INVITE) ←	←	200 OK (INVITE)
		ACK →	→	ACK
NOTIFY 4	←	NOTIFY 4		
200 OK (NOTIFY 4)	→	200 OK (NOTIFY 4)		
BYE	→		→	BYE
200 OK (BYE)	←		←	200 OK (BYE)

SSS__XXSSCONF_ CRE_005	CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_CRE	
Configuration:	CONF	
Selection criteria:	Conference creation by Three-way session creation. REFER request to the focus, Replaces method is used, Conference event package subscribed.	
Test purpose:	**Creation of the conference** Ensure that, when User A sends an INVITE request with request URI set to a valid *conference factory URI*: • User A receives a 200 OK SIP response from the *conference focus* containing "isfocus" feature parameter in Contact header. User A shall store the content of the receive Contact header as the *conference URI*. • User A sends an ACK SIP request. • User A sends a SUBSCRIBE request with request URI set to the *conference URI* (previously stored) and the Event header set to "conference". • User A receives a 200 OK SIP response to the SUBSCRIBE request. • User A receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference". • User A sends a 200 OK SIP response to the NOTIFY request. **Inviting users to the conference** For each active SIP session, User A sends a REFER request to the conference focus with request URI set to the *conference URI* previously stored and Refer-To header set to the SIP URI of the remote user. Also, into the Refer-to header **the replaces method is used** in order to terminate the active SIP session between the user A and the remote user: • User A receives a 202 Accepted SIP response to the REFER request. • Remote user receives an INVITE request from the conference focus to be invited to the conference. The INVITE contains the Replaces header with SIP dialog data ("Call-ID", "From" tag, "To" tag) to be replaced. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A sends a 200 OK SIP response to the NOTIFY request. • Remote user sends a 200 OK SIP response to the INVITE request from the conference focus. • Remote user receives an ACK from the conference focus. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK**. • User A sends a 200 OK SIP response to the NOTIFY request. • Remote user sends a BYE request to the User A in order to release the active SIP session between the user A and the remote user. • User A receives a BYE request from remote user. • User A sends a 200 OK SIP response to the BYE request. • Remote user receives a 200 OK SIP response to the BYE request. • User A receives a NOTIFY from the conference focus (on the same dialog of the SUBSCRIBE previously sent). • User A sends a 200 OK SIP response to the NOTIFY request.	

SSS__XXSSCONF_ CRE_005	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3**	
Precondition:	• User A was participating in two SIP sessions (one with User B and the other with User C). • The SIP session between User A and User B was previously put on HOLD by User A.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** INVITE: Request URI contains the conference factory URI. 200 OK: "isfocus" feature parameter indicated in Contact header field conference URI contains in the Contact header field. NOTIFY: Event header contains conference; Subscription-State header contains active, application/conference-info+xml contains connected, dialled-in REFER 1: Refer-to header contains the SIP URI of the **UA B**. Refer-To: <sip: URI-B?Replaces=call-id1%3Bto-tagsession1%3Bfrom-tagSession1; method=INVITE>. NOTIFY 1: Event header contains **refer**; Subscription-State header contains **active,** Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 2: Event header contains **refer**; Subscription-State header contains **terminated,** Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.** NOTIFY 3: Event header contains **conference**; Subscription-State header contains **active,** application/conference-info+xml contains connected, dialled-out. BYE 1: Call-ID: call-id1/ To: ….; tag=session1/ From: …;tag=Session1. REFER 2: Refer-to header contains the SIP URI of the **UA C**and Replaces header for session 2. Refer-To: <sip: URI-C?Replaces=call-id2%3Bto-tagsession2%3Bfrom-tagSession2; method=INVITE>. NOTIFY 4: Event header contains **refer**; Subscription-State header contains **active,** Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 5: Event header contains **refer**; Subscription-State header contains **terminated,** Content-Type header contains "message/sipfrag", message/sipfrag contains **SIP/2.0 200 OK.** NOTIFY 6: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml contains connected, dialled-out. BYE 2: Call-ID: call-id2/ To: ….; tag=session2/ From: …;tag=Session2.	

Comments:

SIP UA A		**Focus**		**SIP UA B**		**SIP UA C**
		Establishment of session #1				
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
200 OK (INVITE)	←		←	200 OK (INVITE)		
ACK	→		→	ACK		
INVITE (sendonly)	→		→	INVITE (sendonly)		
200 OK (recvonly)	←		←	200 OK (recvonly)		
ACK	→		→	ACK		
		Establishment of session #2				
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	←				←	200 OK (INVITE)
ACK	→				→	ACK
		Conference creation				
INVITE	→	INVITE				
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→	ACK				
SUBSCRIBE	→	SUBSCRIBE				
200 OK (SUBSCRIBE)	←	200 OK (SUBSCRIBE)				
NOTIFY	←	NOTIFY				
200 OK (NOTIFY)	→	200 OK (NOTIFY)				

SSS__XXSSCONF_ CRE_005		CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3				
		Inviting UA B to the conference				
REFER 1	→	REFER 1				
202 Accepted	←	202 Accepted				
		INVITE 4	→	INVITE 4		
NOTIFY 1	←	NOTIFY 1				
200 OK (NOTIFY 1)	→	200 OK (NOTIFY 1)				
		200 OK (INVITE 4)	←	200 OK (INVITE 4)		
		ACK	→	ACK		
NOTIFY 2	←	NOTIFY 2				
200 OK (NOTIFY 2)	→	200 OK (NOTIFY 2)				
BYE 1	←		←	BYE 1		
200 OK (BYE 1)	→		→	200 OK (BYE 1)		
NOTIFY 3	←	NOTIFY 3				
200 OK (NOTIFY 3)	→	200 OK (NOTIFY 3)				
		Inviting UA C to the conference				
REFER 2	→	REFER 2				
202 Accepted	←	202 Accepted				
		INVITE 5	→		→	INVITE 5
NOTIFY 4	←	NOTIFY 4				
200 OK (NOTIFY 4)	→	200 OK (NOTIFY 4)				
		200 OK (INVITE5)	←		←	200 OK(INVITE 5)
		ACK	→		→	ACK
NOTIFY 5	←	NOTIFY 5				
200 OK (NOTIFY 5)	→	200 OK (NOTIFY 5)				
BYE 2	←				←	BYE 2
200 OK (BYE 2)	→				→	200 OK (BYE 2)
NOTIFY 6	←	NOTIFY 6				
200 OK (NOTIFY 6)	→	200 OK (NOTIFY 6)				

<table>
<tr><th>SSS__XXSSCONF_
CRE_06</th><th>CONF reference to:
TS 124 147 [19], clauses 5.2.1, 5.3.1.3</th><th></th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-SIP-SIP/Supplementary_Services/CONF_CRE</td></tr>
<tr><td>Configuration:</td><td colspan="2">CONF</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">Conference creation by Three-way session creation. REFER request to the focus, Replaces method is used, Conference event package not subscribed.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Creation of the conference
Ensure that, when User A sends an INVITE request with request URI set to a valid conference factory URI:
User A receives a 200 OK SIP response from the conference focus containing "isfocus" feature parameter in Contact header. User A shall store the content of the receive Contact header as the conference URI.
User A sends an ACK SIP request.
Inviting users to the conference
For each active SIP session, User A sends a REFER request to the conference focus with request URI set to the conference URI previously stored and Refer-To header set to the SIP URI of the remote user. Also, into the Refer-to header the replaces method is used in order to terminate the active SIP session between the user A and the remote user:
User A receives a 202 Accepted SIP response to the REFER request.
Remote user receives an INVITE request from the conference focus to be invited to the conference. The INVITE contains the Replaces header with SIP dialog data ("Call-ID", "From" tag, "To" tag) to be replaced.
User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 100 Trying.
User A sends a 200 OK SIP response to the NOTIFY request.
Remote user sends a 200 OK SIP response to the INVITE request from the conference focus.
Remote user receives an ACK from the conference focus.
User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 200 OK.
User A sends a 200 OK SIP response to the NOTIFY request.
Remote user sends a BYE request to the User A in order to release the active SIP session between the user A and the remote user.
User A receives a BYE request from remote user.
User A sends a 200 OK SIP response to the BYE request.
Remote user receives a 200 OK SIP response to the BYE request.</td></tr>
<tr><td>Precondition:</td><td colspan="2">User A was participating in two SIP sessions (one with User B and the other with User C).
The SIP session between User A and User B was previously put on HOLD by User A.</td></tr>
</table>

SSS__XXSSCONF_ CRE_06	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3**	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** INVITE: Request URI contains the conference factory URI. 200 OK: "isfocus" feature parameter indicated in Contact header field conference URI contains in the Contact header field. REFER 1: Refer-to header contains the URI of user#2 and Replaces header for session 1. Refer-To: <sip:User#2?Replaces=Call-ID1%3Bto-tagsession1%3Bfrom-tagSession1; method=INVITE>. INVITE 4: Replaces: Call-ID1; to-tag=to-tagSession1; from-tag=from-tagSession1. NOTIFY 1: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying**. NOTIFY 2: Event header contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK**. BYE 1: Call-ID: call-id1/ To: ….; tag=session1/ From: …;tag=Session1. REFER 2: Refer-to header contains the URI of user#3 and Replaces header for session 2. Refer-To: <sip:User#3?Replaces=Call-ID2%3Bto-tag session2%3Bfrom-tag Session2; method=INVITE>. INVITE 5: Replaces: Call-ID2; to-tag=to-tagSession2; from-tag=from-tagSession2. NOTIFY 3: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying**. NOTIFY 4: Event header contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK**. BYE 2: Call-ID: call-id2/ To: ….; tag=session2/ From: …;tag=Session2.	

Comments:

SIP UA A		**Focus**		**SIP UA B**		**SIP UA C**
		Establishment of session #1				
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
200 OK (INVITE)	←		←	200 OK (INVITE)		
ACK	→		→	ACK		
INVITE (sendonly)	→		→	INVITE (sendonly)		
200 OK (recvonly)	←		←	200 OK (recvonly)		
ACK	→		→	ACK		
		Establishment of session #2				
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	←				←	200 OK (INVITE)
ACK	→				→	ACK
		Conference creation				
INVITE	→	INVITE				
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→	ACK				
		Inviting UA B to the conference				
REFER 1	→	REFER 1				
202 Accepted	←	202 Accepted				
		INVITE 4	→	INVITE 4		
NOTIFY 1	←	NOTIFY 1				
200 OK (NOTIFY 1)	→	200 OK (NOTIFY 1)				
		200 OK (INVITE 4)	←	200 OK (INVITE 4)		
		ACK	→	ACK		
NOTIFY 2	←	NOTIFY 2				
200 OK (NOTIFY 2)	→	200 OK (NOTIFY 2)				
BYE 1	←		←	BYE 1		
200 OK (BYE 1)	→		→	200 OK (BYE 1)		

SSS__XXSSCONF_ CRE_06	CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3				
		Inviting UA C to the conference			
REFER 2	→	REFER			
202 Accepted	←	202 Accepted			
		INVITE 5	→	→	INVITE 5
NOTIFY 3	←	NOTIFY 3			
200 OK (NOTIFY 3)	→	200 OK (NOTIFY 3)			
		200 OK (INVITE 5)	←	←	200 OK (INVITE5)
		ACK	→	→	ACK
NOTIFY 4	←	NOTIFY 4			
200 OK (NOTIFY 4)	→	200 OK (NOTIFY 4)			
BYE 2	←			←	BYE 2
200 OK (BYE 2)	→			→	200 OK (BYE 2)

SSS__XXSSCONF_ CRE_007	CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_CRE	
Configuration:	CONF	
Selection criteria:	Conference creation by SIP URI-list. Conference event package subscribed.	
Test purpose:	Ensure that, when User A sends an INVITE request with "resource-list+xml" body (which contains a SIP URI-list of the participants that User A wants to invite to the conference) and request URI set to a valid *conference factory URI*: • User A receives a 200 OK SIP response from the *conference focus* containing "isfocus" feature parameter in Contact header. User A shall store the content of the receive Contact header as the *conference URI*. • User A sends an ACK SIP request. • User A sends a SUBSCRIBE request with request URI set to the *conference URI* (previously stored) and the Event header set to "conference". • User A receives a 200 OK SIP response to the SUBSCRIBE request. • User A receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference". • User A sends a 200 OK SIP response to the NOTIFY request. • Remote user (User B/User C) receives an INVITE request from the conference focus to be invited to the conference. • Remote user (User B/User C) sends a 180 Ringing SIP response to the INVITE request from the conference focus. • Remote user (User B/User C) sends a 200 OK SIP response to the INVITE request from the conference focus. • Remote user receives an ACK from the conference focus. • User A receives a NOTIFY from the conference focus (on the same dialog of the SUBSCRIBE previously sent). • User A sends a 200 OK SIP response to the NOTIFY request.	
Precondition:		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** INVITE: Request URI contains the *conference factory URI*, Require header contains "recipient-list-invite", Content-Disposition header contains "recipient-list", Content-Type header contains "application/resource-lists+xml" and the resource-lists+xml body contains the SIP URI-list of participants at the conference (according to RFC 5366 [25]). 200 OK: "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. SUBSCRIBE: Request URI contained the *conference URI*. NOTIFY: Event header contains conference; Subscription-State header contains active, application/conference-info+xml contains connected, dialled-in INVITE 2: The P-Asserted-Identity contains the *conference URI*. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of UA A. (This is not mandatory) INVITE 3: The P-Asserted-Identity contains the *conference URI*. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of UA A. (This is not mandatory).	

SSS__XXSSCONF_ CRE_007	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.3**				
Comments:					
SIP UA A		**Focus**		**SIP UA B**	**SIP UA C**
		Conference creation			
INVITE	→	INVITE			
200 OK (INVITE)	←	200 OK (INVITE)			
ACK	→	ACK			
SUBSCRIBE	→	SUBSCRIBE			
200 OK (SUBSCRIBE)	←	200 OK (SUBSCRIBE)			
NOTIFY	←	NOTIFY			
200 OK (NOTIFY)	→	200 OK (NOTIFY)			
		Inviting UA B to the conference			
		INVITE 2	→	INVITE 2	
		180 Ringing	←	180 Ringing	
		200 OK (INVITE 2)	←	200 OK(INVITE2)	
		ACK	→	ACK	
NOTIFY	←	NOTIFY			
200 OK (NOTIFY)	→	200 OK (NOTIFY)			
		Inviting UA C to the conference			
		INVITE 3	→	→	INVITE 3
		180 Ringing	←	←	180 Ringing
		200 OK (INVITE 3)	←	←	200 OK (INVITE 3)
		ACK	→	→	ACK
NOTIFY	←	NOTIFY			
200 OK (NOTIFY)	→	200 OK (NOTIFY)			

SSS__XXSSCONF_ CRE_008	**CONF reference to:** **TS 124 147 [19], clauses 5.2.1, 5.3.1.3**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_CRE	
Configuration:	CONF	
Selection criteria:	Conference creation by SIP URI-list. Conference event package not subscribed.	
Test purpose:	Ensure that, when User A sends an INVITE request with "resource-list+xml" body (which contains a SIP URI-list of the participants that User A wants to invite to the conference) and request URI set to a valid *conference factory URI*: • User A receives a 200 OK SIP response from the *conference focus* containing "isfocus" feature parameter in Contact header. User A shall store the content of the receive Contact header as the *conference URI*. • User A sends an ACK SIP request. • Remote user (User B/User C) receives an INVITE request from the conference focus to be invited to the conference. • Remote user (User B/User C) sends a 180 Ringing SIP response to the INVITE request from the conference focus. • Remote user (User B/User C) sends a 200 OK SIP response to the INVITE request from the conference focus. • Remote user receives an ACK from the conference focus.	
Precondition:		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** INVITE: Request URI contains the *conference factory URI*, Require header contains "recipient-list-invite", Content-Disposition header contains "recipient-list", Content-Type header contains "application/resource-lists+xml" and the resource-lists+xml body contains the SIP URI-list of participants at the conference (according to RFC 5366 [25]). 200 OK: "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. INVITE 2: The P-Asserted-Identity contains the *conference URI*. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of UA A. (This is not mandatory) INVITE 3: The P-Asserted-Identity contains the *conference URI*. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of UA A. (This is not mandatory).	

Comments:

SIP UA A		**Focus**		**SIP UA B**		**SIP UA C**
		Conference creation				
INVITE	→	INVITE				
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→	ACK				
		Inviting UA B to the conference				
		INVITE 2	→	INVITE 2		
		180 Ringing	←	180 Ringing		
		200 OK (INVITE 2)	←	200 OK (INVITE 2)		
		ACK	→	ACK		
		Inviting UA C to the conference				
		INVITE 3	→		→	INVITE 3
		180 Ringing	←		←	180 Ringing
		200 OK (INVITE 3)	←		←	200 OK(INVITE 3)
		ACK	→		→	ACK

SSS__XXSSCONF_ CRE_09	**CONF reference to:** **TS 124 147 [19], clauses 5.2.1, 5.3.1.3**		
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_CRE		
Configuration:	CONF		
Selection criteria:	Unsuccessful. Conference creation with a *conference factory URI* not allocated by the conference focus.		
Test purpose:	Ensure that, when User A sends an INVITE request with request URI set to a not valid *conference factory URI*: • User A receives a *488 Not Acceptable Here* SIP response from the conference focus. • User A sends an ACK SIP request		
Precondition:			
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** INVITE: Request URI contains a conference factory URI not allocated by the conference focus.		
Comments:			
SIP UA A			**Focus**
INVITE		→	INVITE
488 Not Acceptable Here		←	488 Not Acceptable Here
ACK		→	ACK

6.2.7.2 Inviting other users to a conference

<table>
<tr><td>SSS__XXSSCONF_INV_001</td><td>CONF reference to:
TS 124 147 [19], clauses 5.2.1, 5.3.1.4, 5.3.1.5</td><td></td></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-SIP-SIP/Supplementary_Services/CONF_INV</td></tr>
<tr><td>Configuration:</td><td colspan="2">CONF</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">Inviting participant by sending REFER to the conference focus. The conference event package is subscribed.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that, when User A sends a REFER request to the conference focus with request URI set to the conference URI previously stored and Refer-To header set to the SIP URI of the remote user (the parameter "method" set to INVITE in the Refer-To header can be included or omitted):
• User A receives a 202 Accepted SIP response to the REFER request.
• Remote user receives an INVITE request from the conference focus to be invited to the conference.
• User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 100 Trying.
• User A sends a 200 OK SIP response to the NOTIFY request.
• Remote user sends a 180 Ringing SIP response to the INVITE request from the conference focus.
• Remote user sends a 200 OK SIP response to the INVITE request from the conference focus.
• Remote user receives an ACK from the conference focus.
• User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 200 OK.
• User A sends a 200 OK SIP response to the NOTIFY request.
• Remote user sends a SUBSCRIBE request with request URI set to the conference URI (previously stored) and the Event header set to "conference".
• Remote User receives a 200 OK SIP response to the SUBSCRIBE request.
• Remote user receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference".
• Remote user sends a 200 OK SIP response to the NOTIFY request.

Repeat the above steps twice in order to invite to the conference User B (when remote user is UA B) and User C (when remote user is UA C).

When User C has joined the conference:
• User B receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference".
• User B sends a 200 OK SIP response to the NOTIFY request.

NOTE: Additionally, User A may include the Referred-By header to the REFER and set it to his SIP URI.</td></tr>
<tr><td>Precondition:</td><td colspan="2">• User A has created a conference by using a conference factory URI.</td></tr>
</table>

SSS__XXSSCONF_INV_001	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.4, 5.3.1.5**	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** REFER 1: Request URI contains the *conference URI (*previously stored)**.** Refer-To header contains the SIP URI of **UA B**. Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) INVITE 2: The P-Asserted-Identity contains the *conference URI*. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) NOTIFY 1: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body, contains **SIP/2.0 100 Trying.** NOTIFY 2: Event contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.** SUBSCRIBE: Request URI contained the conference URI, Event header contains "conference". NOTIFY 3: Event header contains **conference**; Subscription-State header contains **active**, application/conference-info+xml contains connected, dialled-out. REFER 2: Request URI contained the *conference URI (*previously stored)**.** Refer-To header contains the SIP URI of **UA C**. Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) INVITE 3: The P-Asserted-Identity contains the *conference URI*. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) NOTIFY 4: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body, contains **SIP/2.0 100 Trying.** NOTIFY 5: Event contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.** SUBSCRIBE: Request URI contained the conference URI, Event header contains "conference". NOTIFY 6: Event header contains **conference**; Subscription-State header contains **active**, application/conference-info+xml contains connected, dialled-out. NOTIFY 7: Event header contains **conference**; Subscription-State header contains **active**, application/conference-info+xml contains connected, dialled-out.	

Comments:

SIP UA A		**Focus**		**SIP UA B**	**SIP UA C**
		Conference creation			
INVITE	→	INVITE			
200 OK (INVITE)	←	200 OK (INVITE)			
ACK	→	ACK			
		Inviting UA B to the conference			
REFER 1	→	REFER 1			
202 Accepted	←	202 Accepted			
		INVITE 2	→	INVITE 2	
NOTIFY 1	←	NOTIFY 1			
200 OK (NOTIFY 1)	→	200 OK (NOTIFY 1)			
		180 Ringing	←	180 Ringing	
		200 OK (INVITE 2)	←	200 OK (INVITE 2)	
		ACK	→	ACK	
NOTIFY 2	←	NOTIFY 2			
200 OK (NOTIFY 2)	→	200 OK (NOTIFY 2)			
		SUBSCRIBE	←	SUBSCRIBE	
		200 OK (SUBSCRIBE)	→	200 OK(SUBSCRIBE)	
		NOTIFY 3	→	NOTIFY 3	
		200 OK (NOTIFY 3)	←	200 OK (NOTIFY 3)	

SSS__XXSSCONF_INV_001	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.4, 5.3.1.5**						
			Inviting UA C to the conference				
REFER 2	➔	REFER 2					
202 Accepted	🡐	202 Accepted					
			INVITE 3	➔		➔	INVITE 3
NOTIFY 4	🡐	NOTIFY 4					
200 OK (NOTIFY 4)	➔	200 OK (NOTIFY 4)					
			180 Ringing	🡐		🡐	180 Ringing
			200 OK (INVITE 3)	🡐		🡐	200 OK (INVITE 3)
			ACK	➔		➔	ACK
NOTIFY 5	🡐	NOTIFY 5					
200 OK (NOTIFY 5)	➔	200 OK (NOTIFY 5)					
			SUBSCRIBE	🡐		🡐	SUBSCRIBE
			200 OK (SUBSCRIBE)	➔		➔	200 OK(SUBSCRIBE)
			NOTIFY 6	➔		➔	NOTIFY 6
			200 OK (NOTIFY 6)	🡐		🡐	200 OK (NOTIFY 6)
			NOTIFY 7	➔	NOTIFY 7		
			200 OK (NOTIFY 7)	🡐	200 OK (NOTIFY 7)		

SSS__XXSSCONF_INV_002	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.4, 5.3.1.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_INV	
Configuration:	CONF	
Selection criteria:	Inviting participant by sending REFER to the conference focus. The conference event package is not subscribed.	
Test purpose:	Ensure that, when User A sends a REFER request to the conference focus with request URI set to the *conference URI* previously stored and Refer-To header set to the SIP URI of the remote user (the parameter "method" set to INVITE in the Refer-To header can be included or omitted): • User A receives a 202 Accepted SIP response to the REFER request. • Remote user receives an INVITE request from the conference focus to be invited to the conference. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A sends a 200 OK SIP response to the NOTIFY request. • Remote user sends a 180 Ringing SIP response to the INVITE request from the conference focus. • Remote user sends a 200 OK SIP response to the INVITE request from the conference focus. • Remote user receives an ACK from the conference focus. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK**. • User A sends a 200 OK SIP response to the NOTIFY request. **Repeat the above steps twice in order to invite to the conference User B (when remote user is UA B) and User C (when remote user is UA C).** NOTE: Additionally, User A may include the Referred-By header to the REFER and set it to his SIP URI.	

SSS__XXSSCONF_INV_002	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.4, 5.3.1.5**	
Precondition:	• User A has created a conference by using a *conference factory URI*.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** REFER 1: Request URI contains the *conference URI (*previously stored)**.** Refer-To header contains the URI of **UA B**. Referred-By header contains SIP URI of **UA A**. (This is not mandatory) INVITE 2: The P-Asserted-Identity contains the conference URI. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of **UA A**. (This is not mandatory) NOTIFY 1: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 2: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag contains **SIP/2.0 200 OK.** REFER 2: Request URI contained the *conference URI (*previously stored)**.** Refer-To header contains the URI of **UA C**. Referred-By header contains SIP URI of **UA A**. (This is not mandatory) INVITE 3: The P-Asserted-Identity contains the conference URI. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of **UA A**. (This is not mandatory) NOTIFY 3: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 4: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag contains **SIP/2.0 200 OK.**	

Comments:

SIP UA A		**Focus**		**SIP UA B**		**SIP UA C**
		Conference creation				
INVITE	→	INVITE				
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→	ACK				
		Inviting UA B to the conference				
REFER 1	→	REFER 1				
202 Accepted	←	202 Accepted				
		INVITE 2	→	INVITE 2		
NOTIFY 1	←	NOTIFY 1				
200 OK (NOTIFY 1)	→	200 OK (NOTIFY 1)				
		180 Ringing	←	180 Ringing		
		200 OK (INVITE 2)	←	200 OK (INVITE 2)		
		ACK	→	ACK		
NOTIFY 2	←	NOTIFY 2				
200 OK (NOTIFY 2)	→	200 OK (NOTIFY 2)				
		Inviting UA C to the conference				
REFER 2	→	REFER 2				
202 Accepted	←	202 Accepted				
		INVITE 3	→		→	INVITE 3
NOTIFY 3	←	NOTIFY 3				
200 OK (NOTIFY 3)	→	200 OK (NOTIFY 3)				
		180 Ringing	←		←	180 Ringing
		200 OK (INVITE 3)	←		←	200 OK (INVITE 3)
		ACK	→		→	ACK
NOTIFY 4	←	NOTIFY 4				
200 OK (NOTIFY 4)	→	200 OK (NOTIFY 4)				

SSS__XXSSCONF_ INV_003	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.4, 5.3.1.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_INV	
Configuration:	CONF	
Selection criteria:	Inviting participant by sending REFER to the participant. The conference event package is subscribed.	
Test purpose:	Ensure that, when User A sends a REFER request to the remote user with request URI set to the SIP URI of the remote user and Refer-To header set to the *conference URI* previously stored (the parameter "method" set to INVITE in the Refer-To header can be included or omitted): • Remote user receives a REFER request containing the Refer-To header set to the *conference URI*. • Remote user sends a 202 Accepted SIP response to the REFER request. • User A receives a 202 Accepted SIP response to the REFER request. • Remote user sends an INVITE request with request URI set to *conference URI* to the conference focus. • Remote user sends a NOTIFY request to the User A (on the same dialog of the REFER previously received) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A sends a 200 OK SIP response to the NOTIFY request. • Remote user receives a 200 OK SIP response to the NOTIFY request. • Remote user receives a 200 OK SIP response to the INVITE request from the conference focus. • Remote user sends an ACK to the conference focus. • Remote user sends a NOTIFY request to the User A (on the same dialog of the REFER previously received) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK.** • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK**. • User A sends a 200 OK SIP response to the NOTIFY request. • Remote user sends a SUBSCRIBE request with request URI set to the *conference URI* (previously stored) and the Event header set to "conference". • Remote User receives a 200 OK SIP response to the SUBSCRIBE request. • Remote user receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference". • Remote user sends a 200 OK SIP response to the NOTIFY request. **Repeat the above steps twice in order to invite to the conference User B (when remote user is UA B) and User C (when remote user is UA C).** When User C has joined the conference: • User B receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference". • User B sends a 200 OK SIP response to the NOTIFY request. NOTE: Additionally, User A may include the Referred-By header to the REFER and set it to his SIP URI.	
Precondition:	• User A has created a conference by using a *conference factory URI*.	

SSS__XXSSCONF_ INV_003	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.4, 5.3.1.5**	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** REFER 1: Request URI contains the SIP URI of **UA B** Refer-To header contains the *conference URI (*previously stored)**.** Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) INVITE 2: Request URI contains the *conference URI*. The P-Asserted-Identity contains the URI of **UA B**. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) NOTIFY 1: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body, contains **SIP/2.0 100 Trying.** NOTIFY 2: Event contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.** SUBSCRIBE: Request URI contained the *conference URI*, Event header contains "**conference**". NOTIFY 3: Event header contains **conference**; Subscription-State header contains **active**, application/conference-info+xml contains connected, dialled-out. REFER 2: Request URI contains the SIP URI of **UA C.** Refer-To header contains the *conference URI (*previously stored). Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) INVITE 3: Request URI contains the *conference URI*. The P-Asserted-Identity contains the URI of **UA C**. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) NOTIFY 4: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body, contains **SIP/2.0 100 Trying.** NOTIFY 5: Event contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.** NOTIFY 6: Event header contains **conference**; Subscription-State header contains **active**, application/conference-info+xml contains connected, dialled-out. NOTIFY 7: Event header contains **conference**; Subscription-State header contains **active**, application/conference-info+xml contains connected, dialled-out.	

Comments:

SIP UA A		**Focus**		**SIP UA B**	**SIP UA C**
		Conference creation			
INVITE	➔	INVITE			
200 OK (INVITE)	←	200 OK (INVITE)			
ACK	➔	ACK			
		Inviting UA B to the conference			
REFER 1	➔		➔	REFER 1	
202 Accepted	←		←	202 Accepted	
		INVITE 2	←	INVITE 2	
NOTIFY 1	←		←	NOTIFY 1	
200 OK (NOTIFY 1)	➔		➔	200 OK (NOTIFY 1)	
		200 OK (INVITE 2)	➔	200 OK (INVITE 2)	
		ACK	←	ACK	
NOTIFY 2	←		←	NOTIFY 2	
200 OK (NOTIFY 2)	➔		➔	200 OK (NOTIFY 2)	
		SUBSCRIBE	←	SUBSCRIBE	
		200 OK (SUBSCRIBE)	➔	200 OK (SUBSCRIBE)	
		NOTIFY 3	➔	NOTIFY 3	
		200 OK (NOTIFY 3)	←	200 OK (NOTIFY 3)	

SSS__XXSSCONF_ INV_003		CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.4, 5.3.1.5				
		Inviting UA C to the conference				
REFER 2	→				→	REFER 2
202 Accepted	←				←	202 Accepted
		INVITE 3	←		←	INVITE 3
NOTIFY 4	←				←	NOTIFY 4
200 OK (NOTIFY 4)	→				→	200 OK (NOTIFY 4)
		200 OK (INVITE 3)	→		→	200 OK (INVITE 3)
		ACK	←		←	ACK
NOTIFY 5	←				←	NOTIFY 5
200 OK (NOTIFY 5)	→				→	200 OK (NOTIFY 5)
		SUBSCRIBE	←		←	SUBSCRIBE
		200 OK (SUBSCRIBE)	→		→	200 OK (SUBSCRIBE)
		NOTIFY 6	→		→	NOTIFY 6
		200 OK (NOTIFY 6)	←		←	200 OK (NOTIFY 6)
		NOTIFY 7	→	NOTIFY 7		
		200 OK (NOTIFY 7)	←	200 OK (NOTIFY 7)		

SSS__XXSSCONF_ INV_004	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.4, 5.3.1.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_INV	
Configuration:	CONF	
Selection criteria:	Inviting participant by sending REFER to the participant. The conference event package is not subscribed.	
Test purpose:	Ensure that, when User A sends a REFER request to the remote user with request URI set to the SIP URI of the remote user and Refer-To header set to the *conference URI* previously stored (the parameter "method" set to INVITE in the Refer-To header can be included or omitted): • Remote user receives a REFER request containing the Refer-To header set to the *conference URI*. • Remote user sends a 202 Accepted SIP response to the REFER request. • User A receives a 202 Accepted SIP response to the REFER request. • Remote user sends an INVITE request with request URI set to *conference URI* to the conference focus. • Remote user sends a NOTIFY request to the User A (on the same dialog of the REFER previously received) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A sends a 200 OK SIP response to the NOTIFY request. • Remote user receives a 200 OK SIP response to the NOTIFY request. • Remote user receives a 200 OK SIP response to the INVITE request from the conference focus. • Remote user sends an ACK to the conference focus. • Remote user sends a NOTIFY request to the User A (on the same dialog of the REFER previously received) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK.** • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK**. • User A sends a 200 OK SIP response to the NOTIFY request. **Repeat the above steps twice in order to invite to the conference User B (when remote user is UA B) and User C (when remote user is UA C).** NOTE: Additionally, User A may include the Referred-By header to the REFER and set it to his SIP URI.	
Precondition:	• User A has created a conference by using a *conference factory URI*.	

SSS__XXSSCONF_ INV_004	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.4, 5.3.1.5**	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** REFER 1: Request URI contains the SIP URI of **UA B** Refer-To header contains the *conference URI (*previously stored)**.** Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) INVITE 2: Request URI contains the *conference URI*. The P-Asserted-Identity contains the URI of **UA B**. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) NOTIFY 1: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body, contains **SIP/2.0 100 Trying.** NOTIFY 2: Event contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.** REFER 2: Request URI contains the SIP URI of **UA C.** Refer-To header contains the *conference URI (*previously stored). Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) INVITE 3: Request URI contains the *conference URI*. The P-Asserted-Identity contains the URI of **UA C**. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of **UA A.** (This is not mandatory) NOTIFY 3: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body, contains **SIP/2.0 100 Trying.** NOTIFY 4: Event contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.**	

Comments:

SIP UA A		**Focus**		**SIP UA B**		**SIP UA C**
		Conference creation				
INVITE	→	INVITE				
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→	ACK				
		Inviting UA B to the conference				
REFER 1	→		→	REFER 1		
202 Accepted	←		←	202 Accepted		
		INVITE 2	←	INVITE 2		
NOTIFY 1	←		←	NOTIFY 1		
200 OK (NOTIFY 1)	→		→	200 OK (NOTIFY 1)		
		200 OK (INVITE 2)	→	200 OK (INVITE 2)		
		ACK	←	ACK		
NOTIFY 2	←		←	NOTIFY 2		
200 OK (NOTIFY 2)	→		→	200 OK (NOTIFY 2)		
		Inviting UA C to the conference				
REFER 2	→				→	REFER 2
202 Accepted	←				←	202 Accepted
		INVITE 3	←		←	INVITE 3
NOTIFY 3	←				←	NOTIFY 3
200 OK (NOTIFY 3)	→				→	200 OK (NOTIFY 3)
		200 OK (INVITE 3)	→		→	200 OK (INVITE 3)
		ACK	←		←	ACK
NOTIFY 4	←				←	NOTIFY 4
200 OK (NOTIFY 4)	→				→	200 OK (NOTIFY 4)

<table>
<tr><td>SSS__XXSSCONF_ INV_005</td><td>CONF reference to:
TS 124 147 [19], clauses 5.2.1, 5.3.1.4, 5.3.1.5</td><td></td></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-SIP-SIP/Supplementary_Services/CONF_INV</td></tr>
<tr><td>Configuration:</td><td colspan="2">CONF</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">Unsuccessful. User joining a conference by using a not valid conference URI.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that, when User A sends a REFER request to the User B with request URI set to the SIP URI of the User B and Refer-To header set to the conference URI previously stored (the parameter "method" set to INVITE in the Refer-To header can be included or omitted):
User B receives a REFER request containing the Refer-To header set to the conference URI.
User B sends a 202 Accepted SIP response to the REFER request.
User A receives a 202 Accepted SIP response to the REFER request.
Remote user sends an INVITE request with request URI set to a not valid conference URI to the conference focus.
User B sends a NOTIFY request to the User A (on the same dialog of the REFER previously received) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 100 Trying.
User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 100 Trying.
User A sends a 200 OK SIP response to the NOTIFY request.
User B receives a 200 OK SIP response to the NOTIFY request.
User B receives a 488 Not Acceptable Here SIP response to the INVITE request from the conference focus.
User B sends an ACK to the conference focus.
Remote user sends a NOTIFY request to the User A (on the same dialog of the REFER previously received) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains SIP/2.0 503 Service Unavailable.
User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains 503 Service Unavailable.
User A sends a 200 OK SIP response to the NOTIFY request.
</td></tr>
<tr><td>Precondition:</td><td colspan="2">User A has created a conference by using a conference factory URI.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Dial string parameters options=PIXIT
TYPE_SDP= PIXIT;
SIP header values:
REFER 1: Request URI contains the SIP URI of UA B
Refer-To header contains the conference URI (previously stored).
Referred-By contains SIP or tel URI of UA A. (This is not mandatory)
INVITE 2: URI contained the conference URI not allocated in the conference focus.
The P-Asserted-Identity contains the URI of UA B.
Referred-By contains SIP or tel URI of UA A. (This is not mandatory)
NOTIFY 1: Event header contains refer; Subscription-State header contains active, Content-Type header contains "message/sipfrag", message/sipfrag body, contains SIP/2.0 100 Trying.
NOTIFY 2: Event contains refer; Subscription-State header contains terminated, Content-Type header contains "message/sipfrag", message/sipfrag body contains SIP/2.0 503 Service Unavailable.</td></tr>
<tr><td colspan="3">Comments:</td></tr>
</table>

SIP UA A		Focus		SIP UA B
		Conference creation		
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
		Inviting UA B to the conference		
REFER 1	→		→	REFER 1
202 Accepted	←		←	202 Accepted
		INVITE 2	←	INVITE 2
NOTIFY 1	←		←	NOTIFY 1
200 OK (NOTIFY 1)	→		→	200 OK (NOTIFY 1)
		488 Not Acceptable Here	→	488 Not Acceptable Here
		ACK	←	ACK
NOTIFY 2	←		←	NOTIFY 2
200 OK (NOTIFY 2)	→		→	200 OK (NOTIFY 2)

6.2.7.3 Leaving a conference

SSS__XXSSCONF _LEAV_001	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.6**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_LEAV	
Configuration:	CONF	
Selection criteria:	A participant leaves the conference. The conference event package is subscribed.	
Test purpose:	Ensure that, when User B sends a BYE request (in order to leave the conference) to the conference focus with request URI set to the *conference URI* (previously stored): • User B sends a 200 OK SIP response to the BYE request. • User B receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference" and Subscription-State header set to "terminated". • User B sends a 200 OK SIP response to the NOTIFY request.	
Precondition:	• User A has created a conference by using a *conference factory URI*. • User A has invited User B to the conference. • User B has joined the conference. • User B has subscribed to the conference event package.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** NOTIFY 4: Event header contains **conference**; Subscription-State header contains **terminated**, Content-Type header contains "application/conference-info+xml".	

Comments:

SIP UA A		**Focus**		**SIP UA B**
		Conference creation		
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
		Inviting UA B to the conference		
REFER 1	→		→	REFER 1
202 Accepted	←		←	202 Accepted
		INVITE 2	←	INVITE 2
NOTIFY 1	←		←	NOTIFY 1
200 OK (NOTIFY 1)	→		→	200 OK (NOTIFY 1)
		200 OK (INVITE 2)	→	200 OK (INVITE 2)
		ACK	←	ACK
NOTIFY 2	←		←	NOTIFY 2
200 OK (NOTIFY 2)	→		→	200 OK (NOTIFY 2)
		SUBSCRIBE	←	SUBSCRIBE
		200 OK (SUBSCRIBE)	→	200 OK (SUBSCRIBE)
		NOTIFY 3	→	NOTIFY 3
		200 OK (NOTIFY 3)	←	200 OK (NOTIFY 3)
		Conference communication		
		UA B leaves the conference		
		BYE	←	BYE
		200 OK (BYE)	→	200 OK (BYE)
		NOTIFY 4	→	NOTIFY 4
		200 OK (NOTIFY 4)	←	200 OK (NOTIFY 4)

SSS__XXSSCONF _LEAV_002	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.6**	
TSS reference:	S SIP-SIP-SIP/Supplementary_Services/CONF_LEAV	
Configuration:	CONF	
Selection criteria:	A participant leaves the conference. The conference event package is not subscribed.	
Test purpose:	Ensure that, when User B sends a BYE request (in order to leave the conference) to the conference focus with request URI set to the *conference URI* (previously stored): • User B sends a 200 OK SIP response to the BYE request.	
Precondition:	• User A has created a conference by using a *conference factory URI*. • User A has invited User B to the conference. • User B has joined the conference.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;	
Comments:		

SIP UA A		**Focus**		**SIP UA B**
		Conference creation		
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
		SIP UA B joining the conference		
REFER 1	→		→	REFER 1
202 Accepted	←		←	202 Accepted
		INVITE 2	←	INVITE 2
NOTIFY 1	←		←	NOTIFY 1
200 OK (NOTIFY 1)	→		→	200 OK (NOTIFY 1)
		200 OK (INVITE 2)	→	200 OK (INVITE 2)
		ACK	←	ACK
NOTIFY 2	←		←	NOTIFY 2
200 OK (NOTIFY 2)	→		→	200 OK (NOTIFY 2)
		Conference communication		
		Participant leaves the conference		
		BYE	←	BYE
		200 OK (BYE)	→	200 OK (BYE)

6.2.7.4 Removing a conference participant from a conference

SSS__XXSSCONF_ REMOV_001	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.6**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_REMOV	
Configuration:	CONF	
Selection criteria:	A participant removes another conference participant from the conference. The conference event package is subscribed.	
Test purpose:	Ensure that, when User A sends a REFER request to the conference focus with request URI set to the *conference URI* (previously stored) and Refer-To header set to the SIP URI of User B (the parameter "method" must be set to BYE): • User A receives a 202 Accepted SIP response to the REFER request. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A sends a 200 OK SIP response to the NOTIFY request. • User B receives a BYE request from the conference focus to be removed from the conference. • User B sends a 200 OK SIP response to the BYE request. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK**. • User A sends a 200 OK SIP response to the NOTIFY request. • User A receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference". • User A sends a 200 OK SIP response to the NOTIFY request.	
Precondition:	• User A has created a conference by using a *conference factory URI*. • User A has subscribed to the conference event package. • User A has invited User B to the conference. • User B has joined the conference.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** REFER 2: Request URI contains conference URI (previously stored). Refer-To header contains the URI of UA B; method=BYE. Referred-By header contains SIP URI of UA A. (This is not mandatory) NOTIFY 4: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 5: Event header contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.** NOTIFY 6: Event header contains **conference**; Subscription-State header contains **active**, Content-Type header contains "application/conference-info+xml"**.**	

Comments:

SIP UA A		**Focus**		**SIP UA B**
		Conference creation		
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
SUBSCRIBE	→			
200 OK (SUBSCRIBE)	←			
NOTIFY	←	NOTIFY		
200 OK (NOTIFY)	→	200 OK (NOTIFY)		
		UA B joining the conference		
REFER 1	→		→	REFER 1
202 Accepted	←		←	202 Accepted
		INVITE 2	←	INVITE 2
NOTIFY 1	←		←	NOTIFY 1
200 OK (NOTIFY 1)	→		→	200 OK (NOTIFY 1)
		200 OK (INVITE 2)	→	200 OK (INVITE 2)
		ACK	←	ACK
NOTIFY 2	←		←	NOTIFY 2
200 OK (NOTIFY 2)	→		→	200 OK (NOTIFY 2)
NOTIFY 3	←	NOTIFY 3		
200 OK (NOTIFY 3)	→	200 OK (NOTIFY 3)		

SSS__XXSSCONF_ REMOV_001	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.6**	
	Conference communication	
	UA A removes UA B from the conference	
REFER 2 →	REFER 2	
202 Accepted ←	202 Accepted	
NOTIFY 4 ←	NOTIFY 4	
200 OK (NOTIFY 4) →	200 OK (NOTIFY 4)	
	BYE →	BYE
	200 OK (BYE) ←	200 OK (BYE)
NOTIFY 5 ←	NOTIFY 5	
200 OK (NOTIFY 5) →	200 OK (NOTIFY 5)	
NOTIFY 6 ←	NOTIFY 6	
200 OK (NOTIFY 6) →	200 OK (NOTIFY 6)	

SSS__XXSSCONF_ REMOV_002	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.6**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_REMOV	
Configuration:	CONF	
Selection criteria:	A participant removes another conference participant from the conference. The conference event package is not subscribed.	
Test purpose:	Ensure that, when User A sends a REFER request to the conference focus with request URI set to the *conference URI* (previously stored) and Refer-To header set to the SIP URI of User B (the parameter "method" must be set to BYE): • User A receives a 202 Accepted SIP response to the REFER request. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A sends a 200 OK SIP response to the NOTIFY request. • User B receives a BYE request from the conference focus to be removed from the conference. • User B sends a 200 OK SIP response to the BYE request. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK**. • User A sends a 200 OK SIP response to the NOTIFY request.	
Precondition:	• User A has created a conference by using a *conference factory URI*. • User A has invited User B to the conference. • User B has joined the conference.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** REFER 2: Request URI contains conference URI (previously stored). Refer-To header contains the URI of UA B; method=BYE. Referred-By header contains SIP URI of UA A. (This is not mandatory) NOTIFY 3: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 4: Event header contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.**	

Comments:

SIP UA A		**Focus**		**SIP UA B**
		Conference creation		
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
		UA B joining the conference		
REFER 1	→		→	REFER 1
202 Accepted	←		←	202 Accepted
		INVITE 2	←	INVITE 2
NOTIFY 1	←		←	NOTIFY 1
200 OK (NOTIFY 1)	→		→	200 OK (NOTIFY 1)
		200 OK (INVITE 2)	→	200 OK (INVITE 2)
		ACK	←	ACK
NOTIFY 2	←		←	NOTIFY 2
200 OK (NOTIFY 2)	→		→	200 OK (NOTIFY 2)
		Conference communication		
		UA A removes UA B from the conference		
REFER 2	→	REFER 2		
202 Accepted	←	202 Accepted		
NOTIFY 3	←	NOTIFY 3		
200 OK (NOTIFY 3)	→	200 OK (NOTIFY 3)		
			→	BYE
			←	200 OK (BYE)
NOTIFY 4	←	NOTIFY 4		
200 OK (NOTIFY 4)	→	200 OK (NOTIFY 4)		

SSS__XXSSCONF_ REMOV_003	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.6**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_REMOV
Configuration:	CONF
Selection criteria:	The conference owner releases the entire conference by sending a BYE to the focus. The conference event package is subscribed.
Test purpose:	Ensure that, when User A sends a BYE request to the conference focus with request URI set to the *conference URI* (previously stored): • User A receives a 200 OK SIP response to the BYE request. • User B receives a BYE request from the conference focus to be removed from the conference. • User B sends a 200 OK SIP response to the BYE request. • User A receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference". • User A sends a 200 OK SIP response to the NOTIFY request.
Precondition:	• User A has created a conference by using a *conference factory URI*. • User A has subscribed to the conference event package. • User A has invited User B to the conference. • User B has joined the conference.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** BYE: Request URI contains the conference URI (previously stored). NOTIFY 4: Event header contains **conference**; Subscription-State header contains **active**, Content-Type header contains "application/conference-info+xml"**.**

Comments:

SIP UA A		**Focus**		**SIP UA B**
		Conference creation		
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
SUBSCRIBE	→	SUBSCRIBE		
200 OK (SUBSCRIBE)	←	200 OK (SUBSCRIBE)		
NOTIFY	←	NOTIFY		
200 OK (NOTIFY)	→	200 OK (NOTIFY)		
		UA B joining the conference		
REFER 1	→		→	REFER 1
202 Accepted	←		←	202 Accepted
		INVITE 2	←	INVITE 2
NOTIFY 1	←		←	NOTIFY 1
200 OK (NOTIFY 1)	→		→	200 OK (NOTIFY 1)
		200 OK (INVITE 2)	→	200 OK (INVITE 2)
		ACK	←	ACK
NOTIFY 2	←		←	NOTIFY 2
200 OK (NOTIFY 2)	→		→	200 OK (NOTIFY 2)
NOTIFY 3	←	NOTIFY 3		
200 OK (NOTIFY 3)	→	200 OK (NOTIFY 3)		
		Conference communication		
		UA A releases the entire conference		
BYE	→	BYE		
200 OK (BYE)	←	200 OK (BYE)		
		focus removes UA B from the conference		
			→	BYE
			←	200 OK BYE
NOTIFY 4	←	NOTIFY 4		
200 OK (NOTIFY 4)	→	200 OK (NOTIFY 4)		

<table>
<tr><td>SSS__XXSSCONF_ REMOV_004</td><td>CONF reference to:
TS 124 147 [19], clauses 5.2.1, 5.3.1.6</td><td></td></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-SIP-SIP/Supplementary_Services/CONF_REMOV</td></tr>
<tr><td>Configuration:</td><td colspan="2">CONF</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">The conference owner releases the entire conference by sending a BYE to the focus.
The conference event package is not subscribed.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Ensure that, when User A sends a BYE request to the conference focus with request URI set to the conference URI (previously stored):
• User A receives a 200 OK SIP response to the BYE request.
• User B receives a BYE request from the conference focus to be removed from the conference.
• User B sends a 200 OK SIP response to the BYE request.</td></tr>
<tr><td>Precondition:</td><td colspan="2">• User A has created a conference by using a conference factory URI.
• User A has invited User B to the conference.
• User B has joined the conference.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Dial string parameters options=PIXIT
TYPE_SDP= PIXIT;
SIP header values:
BYE: Request URI contains the conference URI (previously stored).</td></tr>
<tr><td colspan="3">Comments:
<table>
<tr><th>SIP UA A</th><th></th><th>Focus</th><th></th><th>SIP UA B</th></tr>
<tr><td colspan="5">Conference creation</td></tr>
<tr><td>INVITE</td><td>→</td><td>INVITE</td><td></td><td></td></tr>
<tr><td>200 OK (INVITE)</td><td>←</td><td>200 OK (INVITE)</td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td>ACK</td><td></td><td></td></tr>
<tr><td colspan="5">UA B joining the conference</td></tr>
<tr><td>REFER 1</td><td>→</td><td></td><td>→</td><td>REFER 1</td></tr>
<tr><td>202 Accepted</td><td>←</td><td></td><td>←</td><td>202 Accepted</td></tr>
<tr><td></td><td></td><td>INVITE 2</td><td>←</td><td>INVITE 2</td></tr>
<tr><td>NOTIFY 1</td><td>←</td><td></td><td>←</td><td>NOTIFY 1</td></tr>
<tr><td>200 OK (NOTIFY 1)</td><td>→</td><td></td><td>→</td><td>200 OK (NOTIFY 1)</td></tr>
<tr><td></td><td></td><td>200 OK (INVITE 2)</td><td>→</td><td>200 OK (INVITE 2)</td></tr>
<tr><td></td><td></td><td>ACK</td><td>←</td><td>ACK</td></tr>
<tr><td>NOTIFY 2</td><td>←</td><td></td><td>←</td><td>NOTIFY 2</td></tr>
<tr><td>200 OK (NOTIFY 2)</td><td>→</td><td></td><td>→</td><td>200 OK (NOTIFY 2)</td></tr>
<tr><td colspan="5">Conference communication</td></tr>
<tr><td colspan="5">UA A releases the entire conference</td></tr>
<tr><td>BYE</td><td>→</td><td>BYE</td><td></td><td></td></tr>
<tr><td>200 OK (BYE)</td><td>←</td><td>200 OK (BYE)</td><td></td><td></td></tr>
<tr><td colspan="5">focus removes UA B from the conference</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>BYE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK (BYE)</td></tr>
</table>
</td></tr>
</table>

SSS__XXSSCONF_ REMOV_005	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.6**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_REMOV	
Configuration:	CONF	
Selection criteria:	The conference owner releases the entire conference by sending a REFER to the focus. The conference event package is subscribed.	
Test purpose:	Ensure that, when User A sends a REFER request to the conference focus with request URI set to the *conference URI* (previously stored) and Refer-To header set to the *conference URI* (the parameter "method" must be set to BYE): • User A receives a 202 Accepted SIP response to the REFER request. • User A receives a BYE request from the conference focus to be removed from the conference. • User B receives a BYE request from the conference focus to be removed from the conference. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A sends a 200 OK SIP response to the NOTIFY request. • User A sends a 200 OK SIP response to the BYE request. • User B sends a 200 OK SIP response to the BYE request. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK**. • User A sends a 200 OK SIP response to the NOTIFY request. • User A receives a NOTIFY request (on the same dialog of the SUBSCRIBE previously sent) with the Event header set to "conference". • User A sends a 200 OK SIP response to the NOTIFY request.	
Precondition:	• User A has created a conference by using a *conference factory URI*. • User A has subscribed to the conference event package. • User A has invited User B to the conference. • User B has joined the conference.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** REFER 2: Request URI contains the conference URI (previously stored). Refer-To header contains the conference URI; method=BYE. Referred-By header contains SIP URI of UA A. (This is not mandatory) NOTIFY 4: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 5: Event header contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.** NOTIFY 6: Event header contains **conference**; Subscription-State header contains **terminated**, Content-Type header contains "application/conference-info+xml"**.**	
Comments:		

SIP UA A		**Focus**		**SIP UA B**
		Conference creation		
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK INVITE)		
ACK	→	ACK		
SUBSCRIBE	→	SUBSCRIBE		
200 OK (SUBSCRIBE)	←	200 OK (SUBSCRIBE)		
NOTIFY	←	NOTIFY		
200 OK (NOTIFY)	→	200 OK (NOTIFY)		
		UA B joining the conference		
REFER 1	→		→	REFER 1
202 Accepted	←		←	202 Accepted
		INVITE 2	←	INVITE 2
NOTIFY 1	←		←	NOTIFY 1
200 OK (NOTIFY 1)	→		→	200 OK (NOTIFY 1)
		200 OK (INVITE 2)	→	200 OK (INVITE 2)
		ACK	←	ACK
NOTIFY 2	←		←	NOTIFY 2
200 OK (NOTIFY 2)	→		→	200 OK (NOTIFY 2)
NOTIFY 3	←	NOTIFY 3		
200 OK (NOTIFY 3)	→	200 OK (NOTIFY 3)		

SSS__XXSSCONF_ REMOV_005	CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.6			
	Conference communication			
	UA A releases the entire conference			
REFER 2	→	REFER 2		
202 Accepted	←	202 Accepted		
BYE	←	BYE		
		BYE	→	BYE
NOTIFY 4	←	NOTIFY 4		
200 OK (NOTIFY 4)	→	200 OK NOTIFY 4		
200 OK (BYE)	→	200 OK (BYE)		
		200 OK (BYE)	←	200 OK (BYE)
NOTIFY 5	←	NOTIFY 5		
200 OK (NOTIFY 5)	→	200 OK (NOTIFY 5)		
NOTIFY 6	←	NOTIFY 6		
200 OK (NOTIFY 6)	→	200 OK (NOTIFY 6)		

SSS__XXSSCONF_ REMOV_006	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.6**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CONF_REMOV
Configuration:	CONF
Selection criteria:	The conference owner releases the entire conference by sending a REFER to the focus. The conference event package is not subscribed.
Test purpose:	Ensure that, when User A sends a REFER request to the conference focus with request URI set to the *conference URI* (previously stored) and Refer-To header set to the *conference URI* (the parameter "method" must be set to BYE): • User A receives a 202 Accepted SIP response to the REFER request. • User A receives a BYE request from the conference focus to be removed from the conference. • User B receives a BYE request from the conference focus to be removed from the conference. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 100 Trying.** • User A sends a 200 OK SIP response to the NOTIFY request. • User A sends a 200 OK SIP response to the BYE request. • User B sends a 200 OK SIP response to the BYE request. • User A receives a NOTIFY (on the same dialog of the REFER previously sent) with the Event header set to "refer" and the Content-Type header set to "message/sipfrag". The message/sipfrag body contains **SIP/2.0 200 OK**. • User A sends a 200 OK SIP response to the NOTIFY request.
Precondition:	• User A has created a conference by using a *conference factory URI*. • User A has invited User B to the conference. • User B has joined the conference.
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; **SIP header values:** REFER 2: Request URI contains the conference URI (previously stored). Refer-To header contains the conference URI; method=BYE. Referred-By header contains SIP URI of UA A. (This is not mandatory) NOTIFY 3: Event header contains **refer**; Subscription-State header contains **active**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 100 Trying.** NOTIFY 4: Event header contains **refer**; Subscription-State header contains **terminated**, Content-Type header contains "message/sipfrag", message/sipfrag body contains **SIP/2.0 200 OK.**

SSS__XXSSCONF_ REMOV_006	**CONF reference to: TS 124 147 [19], clauses 5.2.1, 5.3.1.6**	

Comments:

SIP UA A		**Focus**		**SIP UA B**
		Conference creation		
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
		UA B joining the conference		
REFER 1	→		→	REFER 1
202 Accepted	←		←	202 Accepted
		INVITE 2	←	INVITE 2
NOTIFY 1	←		←	NOTIFY 1
200 OK (NOTIFY 1)	→		→	200 OK (NOTIFY 1)
		200 OK (INVITE 2)	→	200 OK (INVITE 2)
		ACK	←	ACK
NOTIFY 2	←		←	NOTIFY 2
200 OK (NOTIFY 2)	→		→	200 OK (NOTIFY 2)
		Conference communication		
		UA A releases the entire conference		
REFER 2	→	REFER 2		
202 Accepted	←	202 Accepted		
BYE	←	BYE		
		BYE	→	BYE
NOTIFY 3	←	NOTIFY 3		
200 OK (NOTIFY 3)	→	200 OK (NOTIFY 3)		
200 OK (BYE)	→	200 OK (BYE)		
		200 OK (BYE)	←	200 OK (BYE)
NOTIFY 4	←	NOTIFY 4		
200 OK (NOTIFY 4)	→	200 OK (NOTIFY 4)		

6.2.8 Test purposes for Call Waiting

SS___XXSSCW01	**CW reference to: TS 124 615 [21], clause 4.5.5.2**
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CW
Configuration:	The **user B** has subscribed to CW
Selection criteria:	CW and approaching NDUB condition supported, NDUB status can be achieved for user B.
Test purpose:	Ensure that the SUT, when user A sends an INVITE towards user B which is in the approaching NDUB condition, delivers the INVITE to user B containing a Content-Type header set to application/vnd.3gpp.cw+xml and containing a MIME body including a "call-waiting-indication" element.
SIP Parameter values:	INVITE1 Dial string parameters options=PIXIT TYPE_SDP= PIXIT INVITE2 Content-Type header application/vnd.3gpp.cw+xml MIME body with "call-waiting-indication" element

Comments:

SIP UA A		**SUT**		**SIP UA B**
		UA B enters NDUB condition (e.g. by establishing a communication)		
INVITE1	→		→	INVITE2

SS___XXSSCW02	**CW reference to: TS 124 615 [21], clause 4.5.5.2**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CW	
Configuration:	The **user B** has subscribed to CW	
Selection criteria:	CW and approaching NDUB condition supported, NDUB status can be achieved for user B.	
Test purpose:	Ensure that the SUT, having delivered an INVITE indicating Call Waiting to user B which is in the approaching NDUB condition, on receipt of a 415 Unsupported Media Type from user B, sends a 486 Busy Here to user A.	
SIP Parameter values:	INVITE1 Dial string parameters options=PIXIT TYPE_SDP= PIXIT INVITE2 Content-Type header application/vnd.3gpp.cw+xml MIME body with "call-waiting-indication" element	
Comments:		
SIP UA A	**SUT**	**SIP UA B**
	UA B enters NDUB condition (e.g. by establishing a communication)	
INVITE1	→ →	INVITE2
	←	415 Unsupported Media Type
486 Busy Here	← →	ACK
ACK	→	

SS___XXSSCW03	**CW reference to: TS 124 615 [21], clause 4.5.5.2**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CW	
Configuration:	The **user B** has subscribed to CW	
Selection criteria:	CW supported, Notification of calling user of CW status is supported.	
Test purpose:	Ensure that the SUT, having delivered an INVITE from user A to user B, on receipt of a 180 Ringing containing an Alert-Info header set to "urn:alert:service:call-waiting", delivers this 180 Ringing to user A and provides an announcement about the CW condition.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT 180 Ringing Alert-Info header set to "urn:alert:service:call-waiting"	
Comments:		
SIP UA A	**SUT**	**SIP UA B**
INVITE	→ →	INVITE
180 Ringing	← ←	180 Ringing
	Announcement to UE A	

SS___XXSSCW04	**CW reference to: TS 124 615 [21], clause 4.5.5.3**		
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CW		
Configuration:	The **user B** has subscribed to CW		
Selection criteria:	CW and approaching NDUB condition supported, NDUB status can be achieved for user B.		
Test purpose:	Ensure that the SUT, having delivered an INVITE from user B, which is in the approaching NDUB condition, to user A containing a Content-Type header set to application/vnd.3gpp.cw+xml, when user A leaves the NDUB condition and accepts the waiting call, handles the call with normal establishment procedures. Ensure that the voice/data transfer on the media channels is performed correctly (e.g. testing QoS parameters).		
SIP Parameter values:	INVITE1 Dial string parameters options=PIXIT TYPE_SDP= PIXIT INVITE2 Content-Type header application/vnd.3gpp.cw+xml MIME body with "call-waiting-indication" element		
Comments:			
SIP UA A	**SUT**		**SIP UA B**
	UA B enters approaching **NDUB condition (e.g. by establishing a communication)**		
INVITE1	→	→	INVITE2
	UA B leaves approaching **NDUB condition (e.g. by releasing a communication)**		
180 Ringing	←	←	180 Ringing
200 OK INVITE	←	←	200 OK INVITE
ACK	→	→	ACK
	Check media		
BYE	→	→	BYE
200 OK BYE	←	←	200 OK BYE

SS___XXSSCW05	**CW reference to: TS 124 615 [21], clause 4.5.5.3**		
TSS reference:	SIP-SIP-SIP/Supplementary_Services/CW		
Configuration:	The **user B** has subscribed to CW		
Selection criteria:	CW supported, Notification of calling user of CW status is supported.		
Test purpose:	Ensure that the SUT, having delivered a 180 Ringing containing an Alert-Info header set to "urn:alert:service:call-waiting" from user B to user A, when user B accepts the call by sending a 200OK, handles the call with normal establishment procedures. Ensure that the voice/data transfer on the media channels is performed correctly (e.g. testing QoS parameters).		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT 180 Ringing Alert-Info header set to "urn:alert:service:call-waiting"		
Comments:			
SIP UA A	**SUT**		**SIP UA B**
INVITE	→	→	INVITE
180 Ringing	←	←	180 Ringing
	Announcement to UE A		
200 OK INVITE	←	←	200 OK INVITE
ACK	→	→	ACK
	Check media		
BYE	→	→	BYE
200 OK BYE	←	←	200 OK BYE

6.2.9 Test purposes for Completion of Communications to Busy Subscriber

NOTE: The descriptions of invocation and operation of the CCBS service by the communication originating user are not yet fully described in TS 124 642 [22]. Therefore no test purposes have been defined for the current version of this document.

6.2.10 Test purposes for Completion of Communications by No Reply

NOTE: The descriptions of invocation and operation of the CCNR service by the communication originating user are not yet fully described in TS 124 642 [22]. Therefore no test purposes have been defined for the current version of this document.

6.2.11 Test purposes for Explicit Communication Transfer

NOTE: In this clause the following conventions apply:

- user A: transferee, user B: transferor (served user), user C: transfer target.

SSS__XXSSECT01	**ECT reference to: TS 124 529 [23], clause 4.5.2**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/ECTD	
Configuration:	The **user B** has subscribed to ECT	
Selection criteria:	ECT supported.	
Test purpose:	*Blind/Assured transfer, served **user B is callee** in original communication* Ensure that the SUT, when user B has established an original communication with user A and user B requests transfer of the communication towards user C by sending a REFER request to user A: • delivers the REFER request to user A containing the ECT Session Identifier URI and when user A responds with a 202 Accepted and a NOTIFY indicating 100 Trying • delivers the 202 Accepted and the NOTIFY to user B and when user A has held the original communication and sends a new INVITE to the ECT Session Identifier URI • delivers the INVITE to user C and continues normal call establishment between user A and user C and when user A sends a NOTIFY indicating 200 OK to user B and user B sends a BYE to release the original communication. • delivers the BYE to user A and continues normal call release between user A and user B. Ensure that the voice/data transfer on the media channels of the transferred call (A-C) is performed correctly (e.g. testing QoS parameters).	
SIP Parameter values:	REFER1 Request URI: contact URI of user A from original call Refer-To: public address of user C Referred-By: user B REFER2 Request URI: user A Refer-To: ECT Session Identifier Referred-By: user B NOTIFY1 body: 100 Trying INVITE1 Request URI: ECT Session Identifier INVITE2 Referred-By: user B NOTIFY2 body: 20OK	

SSS__XXSSECT01	**ECT reference to: TS 124 529 [23], clause 4.5.2**	

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
		Original communication is established from user A to user B				
REFER2	←		←	REFER1		
202 Accepted	→		→	202 Accepted		
NOTIFY1	→		→	NOTIFY1		
200 OK NOTIFY	←		←	200 OK NOTIFY		

NOTE: TS 124 529 [23] 4.5.2.5 does not specify the order of events, holding of the original communication A-B could also take place before answering to the REFER request.

Re-INVITE (sendonly)	→		→	Re-INVITE (sendonly)		
200 OK INVITE (recvonly)	←		←	200 OK INVITE(recvonly)		
ACK	→		→	ACK		
INVITE1	→				→	INVITE2
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
NOTIFY2	→		→	NOTIFY2		
200 OK NOTIFY	←		←	200 OK NOTIFY		
BYE	←		←	BYE		
200 OK BYE	→		→	200 OK BYE		
		Check media (A-C)				
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

SSS__XXSSECT02	**ECT reference to: TS 124 529 [23], clause 4.5.2**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/ECTD	
Configuration:	The **user B** has subscribed to ECT	
Selection criteria:	ECT supported.	
Test purpose:	*Blind/Assured transfer, served **user B is caller** in original communication* Ensure that the SUT, when user B has established an original communication with user A and user B requests transfer of the communication towards user C by sending a REFER request to user A, • delivers the REFER request to user A containing the ECT Session Identifier URI and when user A responds with a 202 Accepted and a NOTIFY indicating 100 Trying • delivers the 202 Accepted and the NOTIFY to user B and when user A has held the original communication and sends a new INVITE to the ECT Session Identifier URI • delivers the INVITE to user C and continues normal call establishment between user A and user C and when user A sends a NOTIFY indicating 200 OK to user B and user B sends a BYE to release the original communication • delivers the BYE to user A and continues normal call release between user A and user B. Ensure that the voice/data transfer on the media channels of the transferred call (A-C) is performed correctly (e.g. testing QoS parameters).	

SSS__XXSSECT02	**ECT reference to: TS 124 529 [23], clause 4.5.2**	
SIP Parameter values:	REFER1 Request URI: contact URI of user A from original call Refer-To: public address of user C Referred-By: user B REFER2 Request URI: user A Refer-To: ECT Session Identifier Referred-By: user B NOTIFY1 body: 100 Trying INVITE1 Request URI: ECT Session Identifier INVITE2 Referred-By: user B NOTIFY2 body: 20OK	
Comments:		

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
		Original communication is established from user B to user A				
REFER2	←		←	REFER1		
202 Accepted	→		→	202 Accepted		
NOTIFY1	→		→	NOTIFY1		
200 OK NOTIFY	←		←	200 OK NOTIFY		
NOTE: TS 124 529 [23], clause 4.5.2.5 does not specify the order of events, holding of the original communication A-B could also take place before answering to the REFER request.						
Re-INVITE (sendonly)	→		→	Re-INVITE (sendonly)		
200 OK INVITE (recvonly)	←		←	200 OK INVITE(recvonly)		
ACK	→		→	ACK		
INVITE1	→				→	INVITE2
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
NOTIFY2	→		→	NOTIFY2		
200 OK NOTIFY	←		←	200 OK NOTIFY		
BYE	←		←	BYE		
200 OK BYE	→		→	200 OK BYE		
		Check media (A-C)				
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

<table>
<tr><th>SSS__XXSSECT03</th><th>ECT reference to:
TS 124 529 [23], clause 4.5.2</th><th></th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-SIP-SIP/Supplementary_Services/ECTD</td></tr>
<tr><td>Configuration:</td><td colspan="2">The user B has subscribed to ECT</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">ECT supported.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">Consultative transfer, served user B is callee in original communication
Ensure that the SUT, when user A has established an original communication with user B, user B has established a consultation communication with user C and user B requests transfer of the original communication towards user C by sending a REFER request to user A:
delivers the REFER request to user A containing the ECT Session Identifier URI and the call replacement data
and when user A responds with a 202 Accepted and a NOTIFY indicating 100 Trying
delivers the 202 Accepted and the NOTIFY to user A
and when user A has held the original communication and sends a new INVITE to the ECT Session Identifier URI
delivers the INVITE to user C and continues normal call establishment between user A and user C
and when user C sends a BYE to release the consultation communication (B-C)
delivers the BYE to user B and continues normal call release between user C and user B
and when user A sends a NOTIFY indicating 200 OK to user B and user B sends a BYE to release the original communication (A-B)
delivers the BYE to user A and continues normal call release between user A and user B.
Ensure that the voice/data transfer on the media channels of the transferred call (A-C) is performed correctly (e.g. testing QoS parameters).</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">REFER1
Request URI: contact URI of user A from original call
Refer-To: public address of user C, using Replaces: from-tag and to-tag of communication B-C
Referred-By: user B

REFER2
Request URI: user A
Refer-To: ECT Session Identifier
Referred-By: user B

NOTIFY1
body: 100 Trying

INVITE1
Request URI: ECT Session Identifier

INVITE2
Referred-By: user B

NOTIFY2
body: 20OK</td></tr>
</table>

SSS__XXSSECT03	**ECT reference to: TS 124 529 [23], clause 4.5.2**	
Comments:		

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
Original communication is established from user A to user B						
Consultation communication is established from user B to user C						
REFER2	←		←	REFER1		
202 Accepted	→		→	202 Accepted		
NOTIFY1	→		→	NOTIFY1		
200 OK NOTIFY	←		←	200 OK NOTIFY		
NOTE: TS 124 529 [23], clause 4.5.2.5 does not specify the order of events, holding of the original communication A-B could also take place before answering to the REFER request.						
Re-INVITE (sendonly)	→		→	Re-INVITE (sendonly)		
200 OK INVITE (recvonly)	←		←	200 OK INVITE(recvonly)		
ACK	→		→	ACK		
INVITE1	→				→	INVITE2
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
		BYE	←		←	BYE
		200 OK BYE	→		→	200 OK BYE
NOTIFY2	→		→	NOTIFY2		
200 OK NOTIFY	←		←	200 OK NOTIFY		
BYE	←		←	BYE		
200 OK BYE	→		→	200 OK BYE		
Check media (A-C)						
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

SSS__XXSSECT04	**ECT reference to: TS 124 529 [23], clause 4.5.2**	
TSS reference:	SIP-SIP-SIP/Supplementary_Services/ECTD	
Configuration:	The **user B** has subscribed to ECT	
Selection criteria:	ECT supported.	
Test purpose:	*Consultative transfer, served* ***user B is caller*** *in original communication* Ensure that the SUT, when user B has established an original communication with user A, user B has established a consultation communication with user C and user B requests transfer of the original communication towards user C by sending a REFER request to user A: • delivers the REFER request to user A containing the ECT Session Identifier URI and the call replacement data and when user A responds with a 202 Accepted and a NOTIFY indicating 100 Trying • delivers the 202 Accepted and the NOTIFY to user A and when user A has held the original communication and sends a new INVITE to the ECT Session Identifier URI • delivers the INVITE to user C and continues normal call establishment between user A and user C and when user C sends a BYE to release the consultation communication (B-C) • delivers the BYE to user B and continues normal call release between user C and user B and when user A sends a NOTIFY indicating 200 OK to user B and user B sends a BYE to release the original communication (B-A) • delivers the BYE to user A and continues normal call release between user A and user B. Ensure that the voice/data transfer on the media channels of the transferred call (A-C) is performed correctly (e.g. testing QoS parameters).	

SSS__XXSSECT04	**ECT reference to:** **TS 124 529 [23], clause 4.5.2**	
SIP Parameter values:	REFER1 Request URI: contact URI of user A from original call Refer-To: public address of user C, using Replaces: from-tag and to-tag of communication B-CReferred-By: user B REFER2 Request URI: user A Refer-To: ECT Session Identifier Referred-By: user B NOTIFY1 body: 100 Trying INVITE1 Request URI: ECT Session Identifier INVITE2 Referred-By: user B NOTIFY2 body: 20OK	

Comments:

SIP UA A		**SUT**		**SIP UA B**		**SIP UA C**
Original communication is established from user B to user A						
Consultation communication is established from user B to user C						
REFER2	←		←	REFER1		
202 Accepted	→		→	202 Accepted		
NOTIFY1	→		→	NOTIFY1		
200 OK NOTIFY	←		←	200 OK NOTIFY		
NOTE: TS 124 529 [23] 4.5.2.5 does not specify the order of events, holding of the original communication A-B could also take place before answering to the REFER request.						
Re-INVITE (sendonly)	→		→	Re-INVITE (sendonly)		
200 OK INVITE (recvonly)	←		←	200 OK INVITE(recvonly)		
ACK	→		→	ACK		
INVITE1	→				→	INVITE2
180 Ringing	←				←	180 Ringing
200 OK INVITE	←				←	200 OK INVITE
ACK	→				→	ACK
		BYE	←		←	BYE
		200 OK BYE	→		→	200 OK BYE
NOTIFY2	→		→	NOTIFY2		
200 OK NOTIFY	←		←	200 OK NOTIFY		
BYE	←		←	BYE		
200 OK BYE	→		→	200 OK BYE		
Check media (A-C)						
BYE	→				→	BYE
200 OK BYE	←				←	200 OK BYE

Annex A (informative): Bibliography

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- IETF RFC 3515: "The Session Initiation Protocol (SIP) Refer Method".
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- IETF RFC 4967 (2007): "Dial String Parameter for the Session Initiation Protocol Uniform Resource Identifier".
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History

Document history		
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