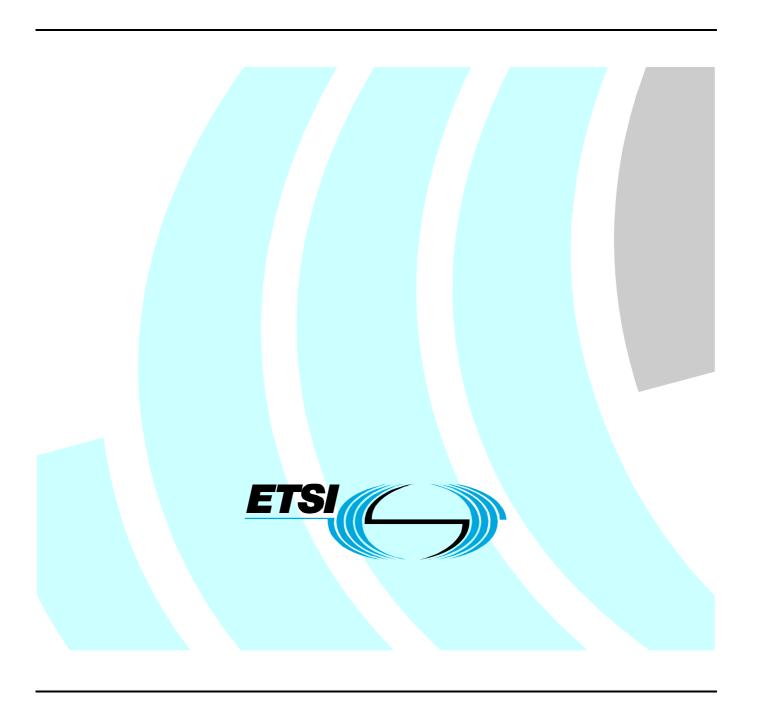
ETSITS 186 001-3 V1.0.0 (2008-04)

Technical Specification

Telecommunications and Internet Converged Services and Protocols for Advanced Networking (TISPAN);
Network Integration Testing between SIP and ISDN/PSTN network signalling protocols;
Part 3: Test Suite Structure and Test Purposes (TSS&TP) for SIP-SIP



Reference DTS/TISPAN-06012-3-NGN-1

Keywords
ISDN, SIP, IP, TSS&TP

ETSI

650 Route des Lucioles F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C Association à but non lucratif enregistrée à la Sous-Préfecture de Grasse (06) N° 7803/88

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN).

The present document is part 3 of a multi-part deliverable covering Network Integration Testing between SIP and ISDN/PSTN network signalling protocols, as identified below:

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

Part 2: "Test Suite Structure and Test Purposes (TSS&TP) ATS and PIXIT";

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

1 Scope

This document specifies the Test Suite Structure and Test Purposes (TSS&TP) for Network Integration Testing (NIT) to verify the overall compatibility of SIP networks. For SIP and SDP specific terminology, reference shall be made to ES 283 003 [1] and RFC 3261 [2] respectively".

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

- [3] Void.
- [4] Void.
- [5] ISO/IEC 9646-1 (1994): "Conformance testing methodology and framework - Part 1: General Concepts".
- [6] ISO/IEC 9646-2 (1994): "Conformance testing methodology and framework - Part 2: Abstract Test Suite Specification".
- [7] ISO/IEC 9646-3 (1992): "Conformance testing methodology and framework - Part 3: The Tree and Tabular Combined Notation".

- [8] ISO/IEC 9646-3/DAM 1 (1992): "Conformance testing methodology and framework Part 3: The Tree and Tabular Combined Notation; Amendment 1: TTCN extensions".
- [9] ISO/IEC 9646-5 (1994): "Conformance testing methodology and framework Part 5: Requirements on test laboratories and clients for the conformance assessment process".
- [10] ISO/IEC 9646-7 (1994): "Conformance testing methodology and framework Part 7: Implementation Conformance Statement etworking (TISPAN); PSTN/ISDN simulation services: Conference (CONF); Protocol specification".
- [11] ETSI TS 124 229: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229)".
- [12] ETSI TS 134 229-1: "Universal Mobile Telecommunications System (UMTS); Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Part 1: Protocol conformance specification (3GPP TS 34.229-1)".
- [13] Void.
- [14] ETSI EN 300 403-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1) protocol; Signalling network layer for circuit-mode basic call control; Part 1: Protocol specification [ITU-T Recommendation Q.931 (1993), modified]".
- [15] ETSI EN 383 001: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control (BICC) Protocol or ISDN User Part (ISUP) [ITU-T Recommendation Q.1912.5, modified]".
- [16] ETSI ES 283 027: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Endorsement of the SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks [3GPP TS 29.163 (Release 7), modified]".
- [17] ETSI TS 183 004: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services: Communication Diversion (CDIV); Protocol specification".
- [18] ETSI TS 183 028: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Common Basic Communication procedures; Protocol specification".
- [19] ETSI TS 183 007: "Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) PSTN/ISDN Simulation Services".
- [20] ETSI TS 183 005: "Telecommunications and Internet converged Services and Protocols for Advanced IETF RFC 3204 (2001), MIME media types for ISDNand QSIG Objects".
- [21] ETSI TS 183 010: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Signalling Control Protocol; Communication HOLD (HOLD); PSTN/ISDN simulation services".

2.2 Informative references

- [22] IETF RFC 2327 (1998): "SDP: Session Description Protocol".
- [23] IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".
- [24] IETF RFC 3311 (2002): "The Session Initiation Protocol UPDATE Method".

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 [2] and RFC 2327 [22] respectively.

SIP precondition: Indicates the support of the SIP "precondition procedure" as defined in RFC 3312 [23].

Inopportune: specifies a test purpose covering a signalling procedure where an inopportune message (type of message not expected in the IUT current state) is sent to the IUT

syntactically invalid: specifies a test purpose covering a signalling procedure where a valid (expected in the current status of the IUT) but not correctly encoded (unknown or incorrect parameter values) message is sent to the IUT, wSIch shall react correctly and eventually reject the message

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) [5] to [10].

valid: specifies a test purpose covering a signalling procedure where all the messages sent to or received from the IUT are valid (expected in the current status of the IUT) and correctly encoded

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.

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 [2]), 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.

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.

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:

GW GateWay Inopportune I

Implementation Under Test IUT Multipoint Control Unit MCU

MGCF Media Gateway Control Function Manufacturer Specific Information MSI

PDU Protocol Data Unit PER Packed Encoding Rules

PICS Protocol Implementation Conformance Statement **PIXIT** Protocol Implementation eXtra Information for Testing

PSase A: Call setup signalling procedures **PSA PSE** PSase E: Call termination signalling procedures

Registration, Admission and Status **RAS**

RCF Register Confirm REGistration **REG** Register Reject RRJ Register Request RRQ Syntactically invalid

STA **STAtus** Terminal TE TP Test Purpose **TSS** Test Suite Structure **UCF Unregistration ConFirm** User Equipment UE Unregistration ReJect URJ Unregistration ReQuest

V Valid

URQ

4 Test Suite Structure (TSS)

4.1 SIP-SIP

C – Plane / U – Plane			
Basic_Call	Successful		
			SSXXxx
		Codec negotiation	SSCNxx
		UPDATE	SSXX_UP_xx
		Unsuccessful	SSXX_Uxx
Supplementary			
Services		OIP/OIR	SSXXSS_OIPxx
			SSXXSS_OIRxx
		TIP/TIR	SSXXSS_TIPxx
		HOLD	SSXXSS_CHxx
		CFU	SSXXSS_CFUxx
		CFB	SSXXSS_CFBxx
		CFNR	SSXXSS_CFNRxx
		CFNL	SSXXSS_CHFNLxx
		CONF	SSXXSS_CONFxx

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 wSIch 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&9:

__: No Supplementary Services Involved / Successful

_U: No Supplementary Services Involved / Unsuccessful

SS: Supplementary Services Involved

SI: Supplementary Services interaction

SN: Nonsymmetrical Supplementary Services Involved

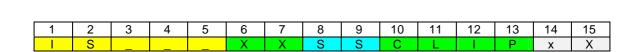
ST: Supplementary Services transparent

5.2 Basic Call

Speech			IS_	XX	<u>_XX</u>					
	2	3	4	5	6	7	8	9	10	11
	_	_			_	_				

5.3 Supplementary Services

CLIP



XXSSCLIP XX

6 Test purposes

The registration and application usage procedures in the ATS shall be compliant to RFC 3261 [2] and ES 283 003 [1] (modified TS 124 229 [11]). The validation of the registration procedure is out of scope of the present document and is part of the Preamble used in the test cases.

The registration conformance tests based on TS 124 229 [11] are contained in TS 134 229-1 [12]

IS

The preconditions mechanism shall be supported by the UE in case of supporting IMS.

The handling of preconditions at the originating or /and terminating UE (MGCF in case if interworking) is described in the following table.

PIXIT Va	alues		
	UE (MGCF) originating case	UE (MGCF) te	rminating case
VA	precondition" option-tag in the Supported header	local resource reservation is required at the terminating UE	local resource reservation is not required by the terminating UE and the terminating UE supports the precondition mechanism
VA_1	"precondition" option-tag in the Supported header	the terminating UE shall make use of the precondition mechanism	
VA_2.1	"precondition" option-tag in the Supported headerd and required resources at the originating network are not reserved	the terminating UE shall make use of the precondition mechanism	
VA_2.2	"precondition" option-tag in the Supported headerd and required resources at the originating network are not reserved		the terminating UE shall use the precondition mechanism
VA_3.1	"precondition" option-tag in the Supported header and required local resources at the originating network	the terminating UE shall make use of the precondition mechanism	
VA_3.2	"precondition" option-tag in the Supported header and required local resources at the originating network		the required local resources at the originating UE and the terminating UE are available, the terminating UE may use the precondition mechanism
VA_4.1	INVITE request does not include the "precondition" option-tag in the Supported header	the terminating UE shall not make use of the precondition mechanism.	
VA_4.2	INVITE request does not include the "precondition" option-tag in the Supported header		the terminating UE shall not make use of the precondition mechanism.

Dial string parameters options

	To header field- UE originated
VA_5.1	sip: dialled digits@homehostportion;user=dialstring
VA_5.2	sip: dialled digits@homehostportion;user=phone
VA_5.3	sip: dialed digits; phone-context=<"+"CC>@homehostportion;user=phone
VA_5.3	sip: dialed digits; phone-context=<"+"CC+NDC>@homehostportion;user=phone

	Request-URI
VA_6.1	E164 Address
	(format "+"CC+NDC+SN)
	(e.g. as User info in SIP URI with user= phone, or as tel URI)

- 6.1 Void
- 6.2 Void
- 6.3 Void
- 6.4 Void
- 6.5 Test purposes for SIP-SIP
- 6.5.1 Test purposes for SIP-SIP, Basic call, Successful

SS XX 01	NGN reference to:					
	RFC 3261 [2]					
	TS 124 229 [11] / ES 283 00	03 [1]				
TSS reference:	SIP-SIP/Basic_call/Success	ful.				
Selection criteria:						
Test purpose:	Ensure that call establishment and the correct handling and mapping of the SDP					
	parameters of the INVITE message defined as : TYPE_SDP is performed correctly.					
	Ensure that in the active call					
			se when the parameter in the			
		is used the codecs in table	1 applies. The call is released			
	from the called user.					
SIP Parameter values:	Dial string parameters option	ns=PIXIT				
	TYPE_SDP= PIXIT;					
	PIXIT for supported header: Case a) No 100 rel;					
	Case b) Supported: 10	00 rol				
	Case c) Supported: 10					
Comments:	A) SDP pre-condition not red					
Comments.	SIP	SUT	SIP			
	INVITE	→ →	INVITE			
	100 Trying	←				
	180 Ringing	+ +	180 Ringing			
	200 OK INVITE	+ +	200 OK INVITE			
	ACK	→ →	ACK			
	BYE	+ +	BYE			
	200 OK BYE	→ →	200 OK BYE			
	C) pre-condition and 100 rel					
	INVITE	→	INVITE			
	100 Trying	+	400 Ci Dr ODD			
		thorize Oos recourse at D				
	Authorize QoS resource at P-SCCF 183 Session Progress SDP ←					
	PRACK	` → →	PRACK			
	TRACK	Resource Reservation at L				
	200 OK PRACK	+ +				
	UPDATE	→ →				
	200 OK UPDATE	+ +	200 OK UPDATE			
	180 Ringing		180 Ringing			
	PRACK	→ →	PRACK			
	200 OK PRACK	+ +				
	200 OK INVITE	-				
	ACK)	,			
	BYE	((-·-			
	200 OK BYE	<u>→</u> →	200 OK BYE			

SSXX02	NGN reference to:				
	RFC 3261 [2]				
	TS 124 229 [11] / ES 283 003 [1]				
TSS reference:	SIP-SIP/Basic_call/Successful.				
Selection criteria:					
Test purpose:	Ensure that call establishment and the correct handling and mapping of the SDP				
	parameters of the INVITE message				
	Ensure that in the active call state th				
	performed correctly (e.g. testing Qo				
	SDP rtpmap: <dynamic-pt> is used</dynamic-pt>	the codecs in table 1	applies. The call is released		
	from the calling user.	-			
SIP Parameter values:	Dial string parameters options=PIXI				
	TYPE_SDP= PIXIT; PIXIT for supported header:				
	Case a) No 100 rel;				
	Case a) No 100 fet, Case b) Supported: 100 rel;				
	Case c) Supported: 100 rel ar	d precondition			
Comments:	Case c) Supported. 100 fer al	a precondition.			
Comments.	A) SDP pre-condition not requested				
	SIP	SUT	SIP		
	INVITE ->	→	INVITE		
	100 Trying	•	1144112		
	180 Ringing	+	180 Ringing		
	200 OK INVITE	÷	200 OK INVITE		
	ACK →	→	ACK		
	BYE →	→	BYE		
	200 OK BYE ←	+	200 OK BYE		
	C) pre-condition and 100 rel				
	INVITE →	→	INVITE		
	100 Trying ←				
		←	183 Session Progress SDP		
		QoS resource at P-S	CCF		
	183 Session Progress SDP ←	_			
	PRACK ->	→	PRACK		
		rce Reservation at UE			
	200 OK PRACK	(200 OK PRACK		
	UPDATE -)	UPDATE		
	200 OK UPDATE ←	(200 OK UPDATE		
	180 Ringing ← PRACK →	←	180 Ringing PRACK		
	PRACK →	→	200 OK PRACK		
	200 OK PRACK	-	200 OK PRACK 200 OK INVITE		
	ACK	→	ACK		
	BYE →	→	BYE		
	200 OK BYE	+	200 OK BYE		
	ZOU ON DIL		ZOO ON DIL		

SSXX03	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]
TSS reference:	SIP-SIP/Basic_call/Successful.
Selection criteria:	
Test purpose:	Ensure that call establishment and the correct if the called user answers with a 180 Ringing message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is performed correctly (e.g. testing QoS parameters). In case when the parameter in the SDP rtpmap: <dynamic-pt> is used the codecs in table 2 applies.</dynamic-pt>
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

SSXX03	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]				
Comments:					
	A) SDP pre-conditi	on not requested	SUT		SIP
	INVITE	→		→	INVITE
	100 Trying	(-	
	180 Ringing	←		←	180 Ringing
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
	BYE	←		←	BYE
	200 OK BYE	→		→	200 OK BYE

SSXX04	NGN reference to: RFC 3261 [2]			
	TS 124 229 [11] / ES 283 003 [1]			
TSS reference:	SIP-SIP/Basic call/Successful.			
Selection criteria:	On -On /Dasic_call/Odocessial.			
Test purpose: Ensure that call establishment and the correctly if the called user answers with with a 200 OK message. In case when the parameter in the SDP rtpmap: <dynamic-pt> is used the codecs in table 2 applies.</dynamic-pt>				
SIP Parameter values:				
Comments:	_			
	A) SDP pre-condition not requested SIP INVITE 100 Trying 200 OK INVITE 4	INVITE		

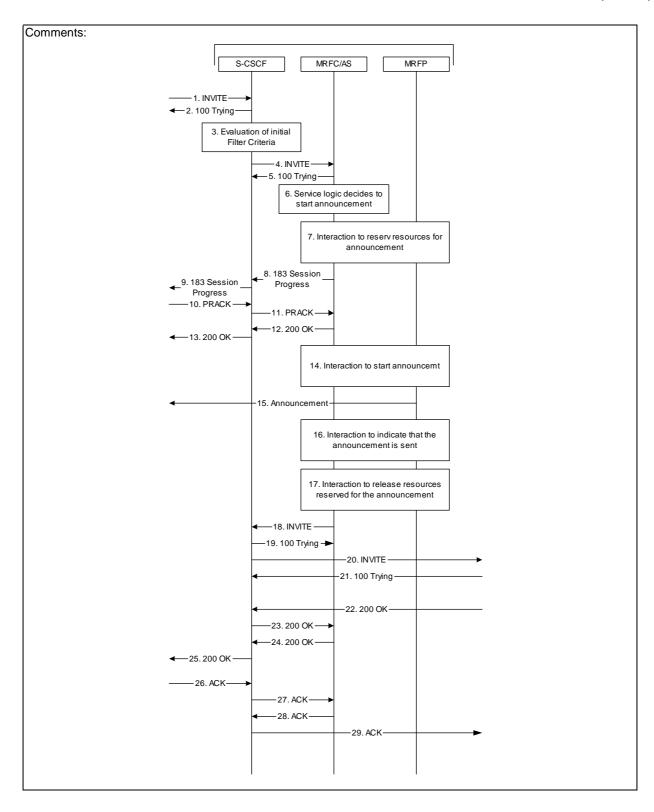
Table 1: Values for the test purpose SS__XX__01 to SS__XX__04

		m= line		b= line	a= line
VA	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwidth-value></bandwidth-value></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters></clock></encoding></dynamic-pt>
				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</dynamic-pt>
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</dynamic-pt>
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

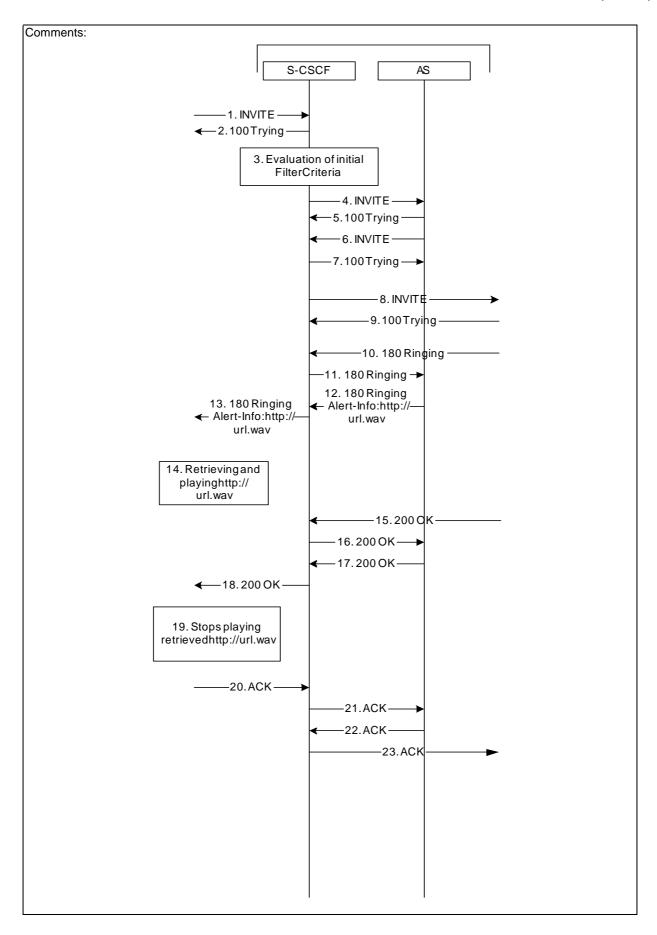
Table 2: Values for test purposes SS___XX__01 and SS___XX__04

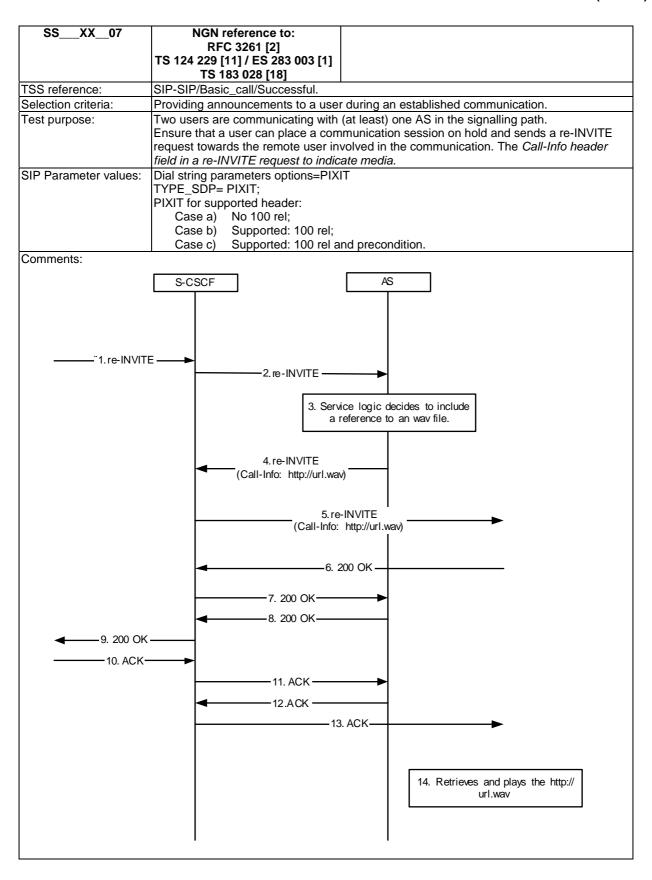
VARIABLE	PT	Encoding	media type	clock rate	channels
VA_01	0	PCMU	Α	8,000	1
VA_02	3	GSM	Α	8,000	1
VA_03	4	G723	Α	8,000	1
VA_04	5	DVI4	Α	8,000	1
VA_05	6	DVI4	Α	16,000	1
VA_06	7	LPC	Α	8,000	1
VA_07	8	PCMA	Α	8,000	1
VA_08	9	G722	Α	8,000	1
VA_09	10	L16	Α	44,100	2
VA_10	11	L16	Α	44,100	1
VA_13	12	QCELP	Α	8,000	1
VA_12	13	CN	Α	8,000	1
VA_13	14	MPA	Α	90,000	
VA_14	15	G728	Α	18,000	1
VA_15	16	DVI4	Α	11,025	1
VA_16	17	DVI4	Α	22,050	1
VA_17	18	G729	Α	8,000	1
VA_18	Dyn	G726-40	Α	8,000	1
VA_19	Dyn	G726-32	Α	8,000	1
VA_20	Dyn	G726-24	Α	8,000	1
VA_21	Dyn	G726-16	Α	8,000	1
VA_22	Dyn	G729D	Α	8,000	1
VA_23	Dyn	G729E	Α	8,000	1
VA_24	Dyn	GSM-EFR	Α	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	

SSXX05	NGN reference to:			
	RFC 3261 [2]			
	TS 124 229 [11] / ES 283 003 [1]			
	TS 183 028 [18]			
TSS reference:	SIP-SIP/Basic_call/Successful			
Selection criteria:				
Test purpose:	Ensure that an AS can send an announcement to the calling user during the			
	establishment of a communication.			
	The AS serving the calling party apply for example when a communication is going to be			
	diverted and the AS serving the diverting user inform the calling party that the			
	communication is going to be diverted.			
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.			



SSXX06	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1] TS 183 028 [18]
TSS reference:	SIP-SIP/Basic_call/Successful
Selection criteria:	Including Alert-Info header field in the 180 (Ringing) response.
Test purpose:	Ensure that according RFC 3261 [2] an Alert-Info header field, to indicate a source of media to play an alternative ring tone by an originating endpoint can be used.
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.





6.5.1.1 Codec negotiation

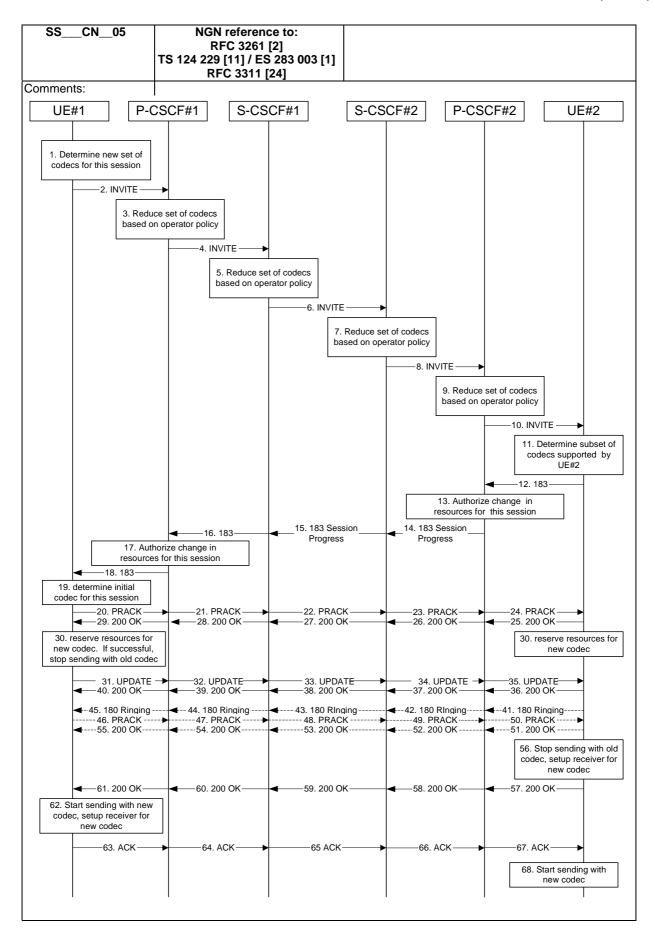
SSCN01	NGN reference to:				
	RFC 3261 [2]				
	TS 124 229 [11] / ES 283 003 [1] RFC 3311 [24]				
TSS reference:	SIP-SIP/Basic_call/Codec negotiation.				
Selection criteria:	On On /Basis_sam/edaes negotiation.				
Test purpose:	During the session, the calling user decides to change the	characteristics of the media			
rest purpose.	During the session, the calling user decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media				
	description. This re-INVITE references the existing dialog so				
	knowsthat it is to modify an existing session instead of estab				
	other party sends a 200 (OK) to accept the change. The req				
	(OK) with an ACK.				
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:	A) SDP pre-condition not requested	015			
	SIP SUT	SIP			
	INVITE And Tradition	INVITE			
	100 Trying ←	400 Dinging			
	180 Ringing ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ←	180 Ringing 200 OK INVITE			
	ACK + +	ACK			
	re-INVITE +	re-INVITE			
	200 OK +	200 OK			
	BYE +	BYE			
	200 OK BYE → →	200 OK BYE			
	C) pre-condition and 100 rel	200 011212			
	INVITE → →	INVITE			
	100 Trying ←				
	+	183 Session Progress			
		SDP			
	Authorize QoS resource at P-SC	CF			
	183 Session Progress				
	SDP				
	PRACK → →	PRACK			
	Resource Reservation at UE1				
	200 OK PRACK ←	200 OK PRACK			
	UPDATE → → + + + + + + + + + + + + + + + + +	UPDATE			
	200 OK UPDATE	200 OK UPDATE 180 Ringing			
	PRACK +	PRACK			
	200 OK PRACK	200 OK PRACK			
	200 OK INVITE	200 OK INVITE			
	ACK → →	ACK			
	re-INVITE →	re-INVITE			
	200 OK ← ←	200 OK			
	BYE ← ←	BYE			
	200 OK BYE → →	200 OK BYE			

SSCN_02	NGN refer	ence to:			
	RFC 32				
	TS 124 229 [11] /	ES 283 003 [1]			
	RFC 331				
TSS reference:	SIP-SIP/Basic_ca	II/Codec negotiati	ion		
Selection criteria:					
Test purpose:			decides to change the		
			ending a re-INVITE con		
	description. This r	e- INVITE referer	nces the existing dialog	so that the	e other party knows
			on instead of establishin		
			ne change.The requesto		
	with an ACK. In case when the parameter in the SDP rtpmap: <dynamic-pt> is used the codecs in table 2 applies.</dynamic-pt>				
CID De remeden velveev			IT.		
SIP Parameter values:	Dial string parame		11		
	PIXIT for supporte				
		100 rel;			
		oported: 100 rel;			
		oported: 100 rel a	and precondition		
Comments:	A) SDP pre-condition				
Commonto.	INVITE	→	<u> </u>	→	INVITE
	100 Trying	-		-	
	180 Ringing	-		(180 Ringing
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
	re-INVITE	←		←	re-INVITE
	200 OK	→		→	200 OK
	BYE	←		(BYE
	200 OK BYE	→		→	200 OK BYE
	C) pre-condition a				
	INVITE	→	→	INVITE	
	100 Trying	←	_		
		Α .1 .	+		sion Progress SDP
	102 Coosian Des		e QoS resource at P-S0	JUF	
	183 Session Prog PRACK	ress SDP ←	→	PRACK	
	PRACK	=	urce Reservation at UE	_	
	200 OK PRACK	₩ ←		200 OK I	PRACK
	UPDATE	÷	÷	UPDATE	-
	200 OK UPDATE	-	-	200 OK I	
	180 Ringing	-	-	180 Ring	
	PRACK	→	→	PRACK	, ,
	200 OK PRACK	←	←	200 OK I	PRACK
	200 OK INVITE	+	+	200 OK I	INVITE
	ACK	→	→	ACK	
	BYE	←	←	BYE	
	re-INVITE	(+	re-INVIT	E
	200 OK	→	→	200 OK	D)/E
	200 OK BYE	<u>→</u>	>	200 OK I	BYE

SSCN03	NGN reference to:		
	RFC 3261 [2]		
	TS 124 229 [11] / ES 283 003 [1]		
	RFC 3311 [24]		
TSS reference:	SIP-SIP/Basic_call/Codec negotiat	ion	
Selection criteria:			
Test purpose:	Ensure that the call establishment	is performed correctl	y.
	 The initial INVITE contains 	a SDP with the offer	1.
			ed in the 180 Ringing message
	Ensure that in the active call state		fer on the media and B-channels
	is performed correctly (e.g. testing		
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT;	IIT	
	PIXIT for supported header:		
	Case a) No 100 rel;		
	Case b) Supported: 100 rel;		
	Case c) Supported: 100 rel a	and precondition.	
Comments:	SDP pre-condition not requested		
	SIP	SUT	SIP
	INVITE offer 1	→ →	INVITE offer1
	100 Trying	(
	180 Ringing Ringing answer 1	(
	200 OK INVITE	=	200 OK INVITE
	ACK	-	ACK
	BYE	-	- BYE
	A) SDP pre-condition not requeste		INIVITE - # 4
	INVITE	→ →	INVITE offer1
	100 Trying	(190 Binging answer 1
	180 Ringing answer 1 200 OK INVITE	-	180 Ringing answer 1200 OK INVITE
	IACK		ACK
	BYE	-	_
	200 OK BYE	→ -	
	C) pre-condition and 100 rel		200 OK BIL
	INVITE offer1	→ -	INVITE offer1
	100 Trying	+	INVITE Offer
	l oo rrying	=	183 Session Progress SDP
	Authoriz	e QoS resource at F	
	183 Session Progress SDP	←	
	PRACK	→ →	PRACK
		urce Reservation at	UE1
	200 OK PRACK	(- 200 OK PRACK
	180 Ringing answer 1	(
	PRACK	→ -	
	200 OK PRACK	+ +	
	200 OK INVITE	+ +	
	ACK	→ -	
	BYE	(
	200 OK BYE	<u>→</u> -	200 OK BYE

SSCN_04	NGN reference to:			
	RFC 3261 [2]			
	TS 124 229 [11] / ES 283 003 [1]			
	RFC 3311 [24]			
TSS reference:	SIP-SIP/Basic_call/Codec negotiati	ion.		
Selection criteria:				
Test purpose:	Ensure that the call establishment			
	The initial INVITE contains			
	 The answer related to the S 	DP offer is contai	ned in the 183 Session Progress	
	message.			
	Ensure that in the active call state (sfer on the media and B-channels	
	is performed correctly (e.g. testing			
SIP Parameter values:	Dial string parameters options=PIX	IT		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;			
	Case c) Supported: 100 rel a			
Comments:	A) SDP pre-condition not requested			
	SIP	SUT	SIP	
	INVITE offer 1	→ →	INVITE offer 1	
	100 Trying	←		
	183 Session Progress answer 1	←	183 Session Progress answer 1	
	180 Ringing	+ +	180 Ringing	
	200 OK INVITE	+ +	200 OK INVITE	
	ACK	> >	ACK	
	BYE	+ +	D D	
	200 OK BYE	→ →	200 OK BYE	
	C) pre-condition and 100 rel			
	INVITE offer 1	→ →	INVITE offer 1	
	100 Trying	←		
		←	183 Session Progress answer 1	
		e QoS resource at	P-SCCF	
	183 Session Progress SDP	←		
	answer 1		DDAGK	
	PRACK		PRACK	
		urce Reservation at		
	200 OK PRACK	+ +	200 OK PRACK	
	180 Ringing PRACK	→ →	180 Ringing PRACK	
	200 OK PRACK	→ ←	200 OK PRACK	
	200 OK PRACK 200 OK INVITE	+ +	200 OK PRACK 200 OK INVITE	
	ACK	→ →	ACK	
	BYE	→ ←	BYE	
	200 OK BYE	→ →	200 OK BYE	
	ZUU OR DIE	, 7	ZUU OR BTE	

SSCN05	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1] RFC 3311 [24]
TSS reference:	SIP-SIP/Basic_call/Codec negotiation.
Selection criteria:	
Test purpose:	Ensure that mobile originated while in home network, establishing a session with another user. After the multimedia session is established, it is possible for either endpoint to change the set of media flows or codec for a media flow.
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.



SSCN_06	NGN reference to:				
	RFC 3261 [2]				
	TS 124 229 [11] / ES 283 003 [1]				
	RFC 3311 [24]				
TSS reference:	SIP-SIP/Basic_call/Codec negotiation.				
Selection criteria:					
Test purpose:	Ensure that the call establishment p				
	 The initial INVITE contains a 	a SDP with the offer 1.			
	Ensure that answer related to the S				
	message. Ensure that in the active call state (N10) the voice transfer on the media and				
	B-channels is performed correctly (ers).		
SIP Parameter values:	Dial string parameters options=PIX	IT			
	TYPE_SDP= PIXIT;				
	PIXIT for supported header:				
	Case a) No 100 rel;				
	Case b) Supported: 100 rel;	1 190			
	Case c) Supported: 100 rel a				
Comments:	A) SDP pre-condition not requested		OLD		
		UT	SIP		
	INVITE offer 1	→	INVITE offer 1		
	100 Trying		100.0 : 0		
	183 Session Progress ← 180 Ringing ←	(183 Session Progress		
	180 Ringing ← 200 OK INVITE answer 1 ←	+	180 Ringing 200 OK INVITE answer 1		
	ACK ->	→			
	BYE	→	7.0		
	200 OK BYE →	,	200 OK BYE		
	200 OR BIL	•	200 OK BTE		
	C) pre-condition and 100 rel				
	/ 1				
	INVITE →	→	INVITE		
	100 Trying ←				
		←	183 Session Progress		
			SDP		
	Authorize QoS resource at P-SCCF	=			
	183 Session Progress ←				
	SDP	_			
	PRACK →	→	PRACK		
	Resource Reservation at UE1	-	000 01/ PD 4 01/		
	200 OK PRACK	(200 OK PRACK		
	180 Ringing	(180 Ringing		
	200 OK INVITE answer 1 ← ACK →	←	200 OK INVITE answer 1		
		→	ACK		
	1	~	BYE		
	200 OK BYE →		200 OK BYE		

SSCN_07	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1] RFC 3311 [24]				
TSS reference:	SIP-SIP/Basic_call/Codec negotiation.				
Selection criteria:					
Test purpose:	 Ensure that the call establishment performed correctly. The initial INVITE contains a SDP with the offer 1. The answer related to the SDP offer is cotained in the 180 Ringing message. A new offer (codec) is sent in the UPDATE message. Ensure that in the active call state (N10) the voice transfer on the media and B-channels is 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:	A) SDP pre-condition not requested				
	SIP SUT SIP INVITE offer 1 → → INVITE offer 1 100 Trying ← + → 183 Session Progress 180 Ringing answer 1 ← ← + → 180 Ringing answer 1 UPDATE offer 2 ← ← UPDATE offer 2 → → 200 OK UPDATE answer 2 → → 200 OK UPDATE answer 2 → → → ACK → → ACK → → ACK → → ACK BYE → → 200 OK BYE → → 200 OK BYE → → 200 OK BYE → → INVITE → → INVITE → INVITE → INVITE → - 183 Session Progress				
	SDP				
	Authorize QoS resource at P-SCCF 183 Session Progress SDP				
	180 Ringing answer 1 ← 180 Ringing answer 1 UPDATE offer 2 ← UPDATE offer 2 200 OK UPDATE answer 2 → 200 OK UPDATE answer 2				
	200 OK INVITE				

SSCN_08	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1] RFC 3311 [24]				
TSS reference:	SIP-SIP/Basic_call/Codec negotiation				
Selection criteria:	on on reason_cam escale negation				
Test purpose:	Ensure that the call establishment i	s performed correctly			
	 Ensure that answer related Progress message. A new of Ensure that in the active can B-channels is performed con 	to the SDP offer is co offer (codec) is sent in Il state (N10) the voice	otained in the 183 Session In the 180 Ringing. The transfer on the media and		
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel a	ΙΤ			
Comments:	A) SDP pre-condition not requested				
	SIP INVITE offer 1	SUT →	SIP INVITE offer 1		
	183 Session Progress answer	+	183 Session Progress answer 1		
	200 OK INVITE ACK answer 2 BYE		5 5		
	C) pre-condition and 100 rel				
	INVITE offer 1 100 Trying ←	→	INVITE offer 1 183 Session Progress answer		
	Authoriz	o Oos roomiroo et D	1		
	183 Session Progress answer 1	e QoS resource at P-	SCCF		
	PRACK →	→	PRACK		
		urce Reservation at U			
	200 OK PRACK 180 Ringing offer 2 PRACK 200 OK PRACK 200 OK INVITE ACK answer 2 BYE 200 OK BYE ←	+ + + +	200 OK PRACK 180 Ringing offer 2 PRACK 200 OK PRACK 200 OK INVITE ACK answer 2 BYE 200 OK BYE		
	BYE ←	←	BYE		

6.5.1.2 UPDATE method

SSXX_UP_01	NGN reference to: RFC 3261 [2]							
		TS 124 229 [11] / ES 283 003 [1]						
T00 (RFC 3311 [24]						
TSS reference:	SIP-SIP/Basic_call/update							
Selection criteria:	UPDATE procedure for the caller.							
Test purpose:	Ensure that if the UPDATE is being							
	transaction, and the initial INVITE							
	offer if the callee generated an ans							
	has received answers to any other		nd has generated answers					
	for any offers it received in an UPD							
SIP Parameter values:	Dial string parameters options=PIX	IT						
	TYPE_SDP= PIXIT;							
	PIXIT for supported header:							
	Case a) No 100 rel;							
	Case b) Supported: 100 rel;							
	Case c) Supported: 100 rel a	ind precondition.						
Comments:	OID.	0.1.7	O.D.					
	SIP	SUT	SIP					
	INVITE offer 1 →	→	INVITE offer 1					
	100 Trying ←	←	183 Session answer1					
	Authoriz	e QoS resource at P-SCC						
		e Qos resource at P-SCC	, F					
	183 Session Progress ← answer 1							
	PRACK	→	PRACK					
	200 OK PRACK	-	200 OK PRACK					
		urce Reservation at UE1	200 OKT KACK					
	UPDATE offer 2 →		UPDATE offer 2					
	200 OK UPDATE	+	200 OK UPDATE					
	180 Ringing answer 2	÷	180 Ringing answer 2					
	PRACK	÷	PRACK					
	200 OK PRACK ←	É	200 OK PRACK					
	200 OK INVITE	-	200 OK INVITE					
	ACK →	→	ACK					
	BYE ←	←	BYE					
	200 OK BYE →	→	200 OK BYE					

SSXX_UP_02	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1] RFC 3311 [24]
TSS reference:	SIP-SIP/Basic_call/update
Selection criteria:	UPDATE procedure for the caller
Test purpose:	Ensure that if the UPDATE is being sent before completion of the initial INVITE transaction, and the initial INVITE did not contain an offer, the UPDATE can contain an offer if the callee generated an offer in a reliable provisional response, and the UAC generated an answer in the corresponding PRACK.
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

SSXX_UP_02	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [7 RFC 3311 [24]	1]			
Comments:					
	SIP		SUT		SIP
	INVITE	→		→	INVITE
	100 Trying	←			
	183 Session offer 1	←		←	183 Session offer 1
	PRACK answer 1	→		→	PRACK answer 1
	200 OK PRACK	←		←	200 OK PRACK
	UPDATE offer 2	→		→	UPDATE offer 2
	200 OK UPDATE with anwer 2	←		←	200 OK UPDATE with anwer 2
	180 Ringing	←		←	180 Ringing
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
	BYE	+		←	BYE

SS XX UP 03	NGN reference	to:				
CCXXCI _CC	RFC 3261 [2]					
	TS 124 229 [11] / ES 2					
	RFC 3261 [2], RFC 3311 [
	RFC 3312 [23	-				
TSS reference:	SIP-SIP/Basic_call/update					
Selection criteria:	UPDATE procedure for the ca	aller.				
Test purpose:	Ensure that if the UPDATE is		e completion	on of the initial INVITE		
' '				er, the UPDATE can contain an		
	offer if the callee generated a					
	generated an answer in the c			,		
SIP Parameter values:	Dial string parameters options					
	TYPE_SDP= PIXIT;					
	PIXIT for supported header:					
	Case a) No 100 rel;					
	Case b) Supported: 100	rel;				
	Case c) Supported: 100	rel and precond	ition			
Comments:						
	SIP	SUT		SIP		
	INVITE	→	→	INVITE		
	100 Trying	←				
	183 Session offer 1	←	←	183 Session offer 1		
	PRACK	→	→	PRACK		
	200 OK	←	←	200 OK		
	UPDATE offer 1	→	→	UPDATE offer 1		
	200 OK UPDATE	←	←	200 OK UPDATE		
	UPDATE offer 2	←	←	UPDATE offer 2		
	200 OK UPDATE	→	→	200 OK UPDATE		
	UPDATE answer 2	→	→	UPDATE answer 2		
	200 OK UPDATE	((200 OK UPDATE		
	180 Ringing ← ← 180 Ringing					
	200 OK INVITE	((200 OK INVITE		
	ACK	→	→	ACK		
	BYE	((BYE		
	200 OK BYE	→	→	200 OK BYE		

SSXX_UP_04	NGN reference to: RFC 3261 [2]		
	TS 124 229 [11] / ES 283 003 [1]		
	RFC 3261 [2],RFC 3311 [24] clause 4,		
	RFC 3312 [23]		
TSS reference:	SIP-SIP/Basic_call/Successful		
Selection criteria:			
Test purpose:	Ensure that when a UAS receives an IN reliable provisional response containing header field that lists the UPDATE methor capable of receiving an UPDATE requesting the containing that it is the UPDATE requesting an UPDATE requesting the containing that is the containing that	SDP, that respor od. This informs	nse SHOULD contain an Allow
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 pr		
Comments:	SIP INVITE SDP1 with allow → header 100 Trying ← 183 SDP2 with allow header PRACK SDP 3 200 OK PRACK ← UPDATE SDP 4 200 OK UPDATE ← 180 Ringing ← 200 OK INVITE ← ACK BYE ← ←	← → ←	200 011 01 27112

SSXX_UP_05	NGN reference to RFC 3261 [2] TS 124 229 [11] / ES 28 RFC 3261 [2],RFC 3311 [2- RFC 3312 [23]	3 003 [1]				
TSS reference:	SIP-SIP/Basic_call/Successfu	l				
Selection criteria:	SHOULD Tests.					
Test purpose:	Ensure that a 2XX response smethod.	SHOULD contain a	n Allow he	eader field listing the UPDATE		
SIP Parameter values:	Dial string parameters options TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 Case c) Supported: 100	rel;	on.			
Comments:	SIP	SUT		SIP		
	INVITE SDP1 with allow header 100 Trying	→		INVITE SDP1 with allow header		
	183 SDP2 with allow header	÷	←	183 SDP2 with allow header		
	PRACK SDP 3	→	→	PRACK SDP 3		
	200 OK PRACK with allow header	←		200 OK PRACK with allow header		
	UPDATE SDP 4	→	→	UPDATE SDP 4		
	200 OK UPDATE ← 200 OK UPDATE					
	180 Ringing					
	200 OK INVITE	← →		200 OK INVITE		
	ACK BYE	7		ACK BYE		
	200 OK BYE	}		200 OK BYE		

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

6.5.1.3.1 Unsuccessful

SSXX_U01	NGN reference to: RFC 3261 [2]				
	TS 124 229 [11] / ES 283 003 [1]				
TSS reference:	SIP-SIP/Basic_call/Unsuccessful				
Selection criteria:					
Test purpose:	1	Ensure that, when calling to unallocated number, the network initiate call clearing to the calling user with a 404 Not Found message.			
SIP Parameter values:	TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel;	PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel;			
Comments:	SIP INVITE →	SUT	SIP INVITE		
	100 Trying ← 404 Not Found ← ACK →	←	404 Not Found ACK		

SSXX_U02	NGN reference to: RFC 3261 [2]		
	TS 124 229 [11] / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful		
Selection criteria:			
Test purpose:	Ensure that the call will be released clearing to the calling user with a 5		available. The network initiates call lable message.
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel a		
Comments:	SIP INVITE → 100 Trying ←	SUT	→ INVITE
	503 Sevice Unawailable		← 503 Sevice Unawailable→ ACK

SSXX_U03		reference to: [2] and RFC 2327			
TSS reference:	CID CID/Posi	[22] c call/Unsuccessful			
	SIF-SIF/Dasi	C_Call/OffSuccessful			
Selection criteria:					
Test purpose:		when the called user i 36 Busy Here messag	•	rk in	itiate call clearing to the calling
SIP Parameter values:	TYPE_SDP= PIXIT for sup Case a) Case b)	rameters options=PIX PIXIT; ported header: No 100 rel; Supported: 100 rel; Supported: 100 rel a			
Comments:	;	SIP	SUT		SIP
	INVITE	→		→	INVITE
	100 Trying	←			
	486 Busy Her	re ←		←	486 Busy Here
	ACK	→		→	ACK

SSXX_U04		reference FC 3261 [2] [11] / ES 2]				
TSS reference:	SIP-SIP/Bas	sic_call/Uns	uccessful				
Selection criteria:							
Test purpose:	from INVITE	messages), the networ	er from the called ork initiate call clea ge or 408 reques	aring to the ca		
SIP Parameter values:	Dial string partype_SDP=PIXIT for supersupersupersupersupersupersupersuper	= PIXIT; pported hea No 100 re Supporte	ader: el; d: 100 rel;	T			
Comments: UE1 INVITE		Proxy 1		Proxy 2			UE2
100 ←	INVITE 100	→ ←	INVITE	→	INVITE INVITE INVITE INVITE INVITE INVITE	> > 	
480 ← ACK →	480 ACK	← →	480 ACK	← →	HAALIE	•	

SSXX_U05	NGN reference to:				
	draft-ietf-sipping-basic-call-				
	flows-02.txt				
TSS reference:	SIP-SIP/Basic_call/Unsuccessful				
Selection criteria:	A transaction timer regarding the c	all establishment is im	plemented at the UAS		
Test purpose:	Ensure that when there is no answ	er from the called use	er, the user initiate call clearing to		
	the calling user with a 480 Tempor	arily unavailable mess	sage (do not disturbe service).		
SIP Parameter values:	Dial string parameters options=PIX	ΊΤ			
	TYPE_SDP= PIXIT;				
	PIXIT for supported header:				
	Case a) No 100 rel;				
	Case b) Supported: 100 rel;				
	Case c) Supported: 100 rel a	nd precondition			
Comments:	SIP UA A	SUT	SIP UA B		
	INVITE →)	INVITE		
	180 Ringing ←	+	 180 Ringing 		
	480 Teporary unavaible ←	+	 480 Teporary unavaible 		
	ACK →	_	ACK		

SSXX_U06	NGN reference to:			
	RFC 3261 [2]			
	TS 124 229 [11] / ES 283 003 [1]			
TSS reference:	SIP-SIP/Basic_call/Unsuccessful			
Selection criteria:				
Test purpose:	Ensure that when the number is changed, the network initiate call clearing to the calling			
	user with a 410 Gone message.			
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.			

SSXX_U06	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]			
Comments:	SIP	SUT		SIP
	INVITE	•	→ INVITE	
	100 Trying			
	410m Gone	_	← 410 Gone	
	ACK -	•	→ ACK	

SSXX_U07	NGN reference to: RFC 3261 [2] TS 124 229 / ES 283 003 [1]	
TSS reference:	SIP-SIP/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that when the destination is calling user with a 502 Bad Gateway	ork initiate call clearing to the
SIP Parameter values:	Dial string parameters options=PIXITYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and	
Comments:	SIP INVITE → 100 Trying ← 502 Bad Gateway ACK →	SIP INVITE 502 Bad Gateway ACK

SSXX_U08	NGN reference to: RFC 3261 [2]		
	TS 124 229 [11] / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful		
Selection criteria:			
Test purpose:	Ensure that the call will be released initiates call clearing to the calling u		
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel are		
Comments:	SIP INVITE 100 Trying 484 Address Inconplete	SUT →	484 Address Inconplete
	ACK →		ACK

SSXX_U09	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]
TSS reference:	SIP-SIP/Basic_call/Unsuccessful
Selection criteria:	
Test purpose:	Ensure that when the called user is not compatible the network initiates call clearing to the calling user with a 503 Service unavailable message.
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.

SSXX_U09	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003	[1]		
Comments:	SIP	5	SUT	SIP
	INVITE	→	→	INVITE
	100 Trying	←		
	503 Service unavailable	←	←	503 Service unavailable
	ACK	→	→	ACK

SSXX_U	J10		reference					
			FC 3261 [2]					
TSS reference:		TS 124 229 SIP-SIP/Bas						
Selection criteria	a:	on on that	<u></u>	accocciai				
Test purpose:						II with a CANC		
SIP Parameter	values:	Dial string pa	Case b) Supported: 100 rel;					The Called user.
Comments:		Case a)						
INIV/ITE	UE1		Proxy 1		Proxy 2		UE2	
INVITE 100 CANCEL	→ ←	INVITE 100	→ ←	INVITE	→	INVITE INVITE INVITE INVITE INVITE INVITE INVITE CANCEL		
				Case h)				
INVITE 100	→	INVITE 100	→ ←	Case b)	→	INVITE INVITE INVITE INVITE INVITE INVITE	+ + + + + + +	
CANCEL	→	CANCEL	→	CANCEL Case c)	→	180 CANCEL	← →	
INVITE 100	→ ←	INVITE 100	→ ←	INVITE	→	INVITE INVITE INVITE INVITE INVITE INVITE INVITE	> + + + + + + + + + + + + + + + + + + +	
CANCEL	→	CANCEL	→	CANCEL	→	200 OK CANCEL	← →	

SSXX_U11	NGN reference to:				
	RFC 3261 [2]				
	TS 124 229 [11] / ES 283 003 [1]				
TSS reference:	SIP-SIP/Basic_call/Unsuccessful				
Selection criteria:					
Test purpose:	During the session, the calling user decide to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re- INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session.				
	Ensutre that if the other party does such as 488 (Not Acceptable Her				
SIP Parameter values:	Dial string parameters options=PITYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel	XIT			
Comments:	SIP	SUP	SIP		
	INVITE -	· →	INVITE		
	100 Trying	i			
	183 Session Progress		183 Session Progress		
	180 Ringing		180 Ringing		
	200 OK INVITE ₩	=	200 OK INVITE		
	ACK -	=	ACK		
	RE-INVITE offer 2 488 Not Acceptable Here BYE 200 OK BYE	← Communication	RE-INVITE offer 2 488 Not Acceptable Here BYE 200 OK BYE		

SSXX_U12	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]				
TSS reference:	SIP-SIP/Basic_call/Unsuccessful				
Selection criteria:					
Test purpose:	During the session, the called user decide to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. Ensutre that if the other party does not accept the change, he sends an error response such as 488 (Not Acceptable Here), which also receives an ACK.				
SIP Parameter values:	Dial string parameters options=PIXI TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and	Т			
Comments:	SIP INVITE 100 Trying 183 Session Progress 180 Ringing 200 OK INVITE ACK RE-INVITE offer 2 488 Not Acceptable Here BYE 200 OK BYE	SUT Communication Communication	INVITE 183 Session Progress 180 Ringing 200 OK INVITE ACK RE-INVITE offer 2 488 Not Acceptable Here BYE 200 OK BYE		

SSXX_U13	NGN reference to: RFC 3261 [2]			
	TS 124 229 [11] / ES 283 00	3 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsucces	sful		
Selection criteria:				
Test purpose:	message. The media	stream is rejected (po	rt nun	
SIP Parameter values:	Ensure that the call is rejected by sending a CANCEL or BYE. 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 offer1 100 Trying 183 Session Progress 180 Ringing answer 1	→ ← ←	→ ← ←	INVITE offer1 183 Session Progress 180 Ringing answer 1
	Case A) CANCEL 200 OK 487 Request Terminated ACK Case B)	← → ← →	+ + + + + + + + + + + + + + + + + + +	CANCEL 200 OK 487 Request Terminated ACK
	BYE 200 OK 487 Request Terminated ACK	← → ← →	+ + + +	BYE 200 OK 487 Request Terminated ACK

SSXX_U14	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful		
Selection criteria:			
Test purpose:	 Ensure that answer related Progress message. The me Ensure that the call is reject 	dia stream is rejected	(port number is set to zero).
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel ar	nd precondition.	
Comments:	SIP INVITE offer1 →	SUT	SIP INVITE offer1
	100 Trying + 183 Session Progress answer 1	←	183 Session Progress answer
	Case A) CANCEL 200 OK 487 Request Terminated ACK CANCEL ← ACK ←	+ + + +	CANCEL 200 OK 487 Request Terminated ACK
	Case B) BYE 200 OK 487 Request Terminated ACK Case B) ← ACK	← → ←	BYE 200 OK 487 Request Terminated ACK

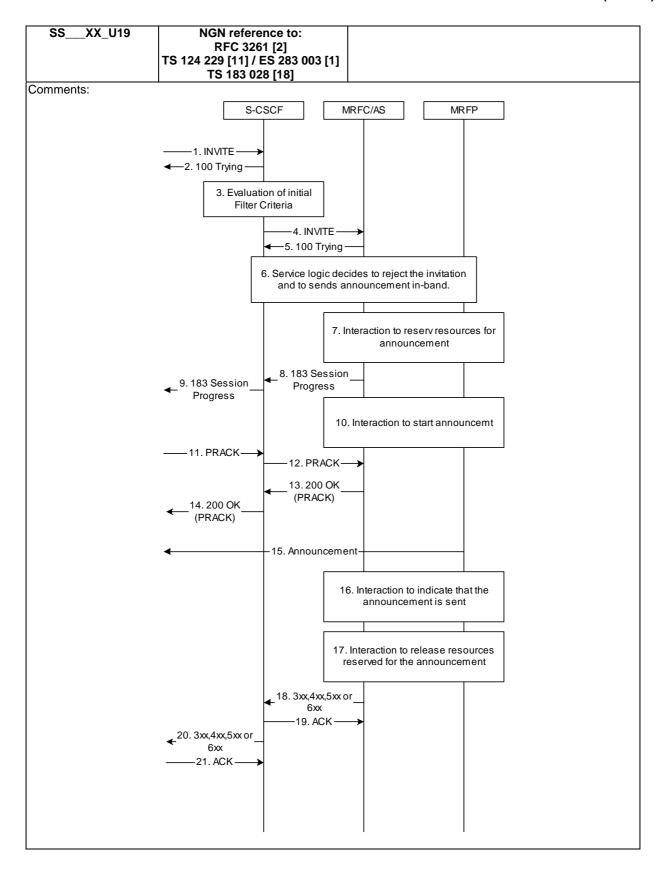
SSXX_U15	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful		
Selection criteria:			
Test purpose:	Ensure that answer related message. The media stream Ensure that the call is reject	n is rejected (port	•
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel ar		
Comments:	SIP INVITE offer1 → 100 Trying ← 200 OK answer 1 → ACK → BYE ← 200 OK →	SUT	SIP → INVITE offer1 → 200 OK answer 1 → ACK ← BYE → 200 OK

SSXX_U16	NGN reference to: RFC 3261 [2]		
	TS 124 229 [11] / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful		
Selection criteria:			
Test purpose:	Ensure that when there is no answer calling user initiate call clearing to the		
SIP Parameter values:	Dial string parameters options=PIXITYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel an		
Comments:	SIP INVITE 100 Trying 180 Ringing CANCEL 200 OK CANCEL 487 Request Terminated ACK	SUT →	180 Ringing CANCEL 200 OK CANCEL 487 Request Terminated

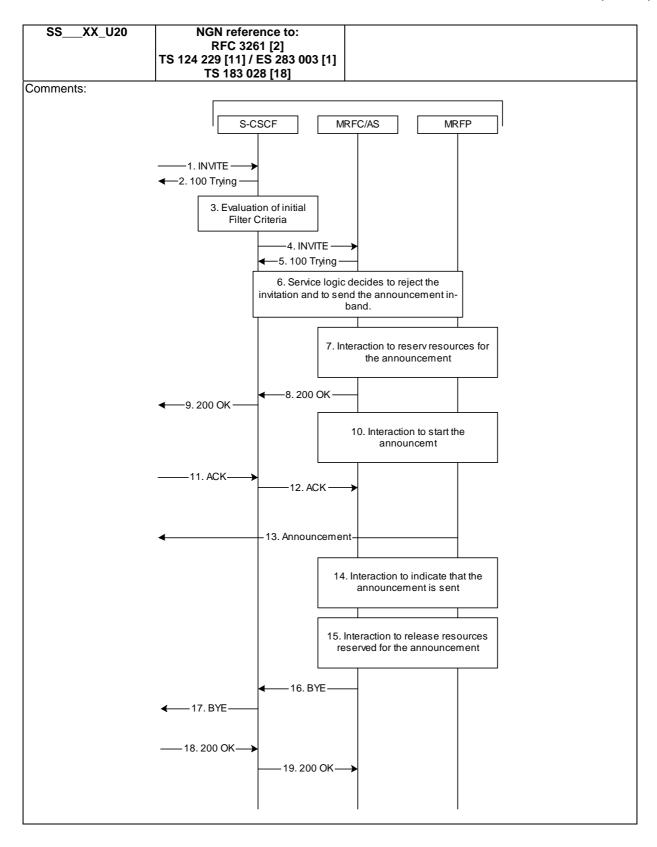
SSXX_U17	NGN reference to: RFC 3261 [2]
	TS 124 229 [11] / ES 283 003 [1]
TSS reference:	SIP-SIP/Basic_call/Unsuccessful
Selection criteria:	
Test purpose:	Ensure that upon receiving a 421 (Extension Required) response to an initial INVITE request in which the precondition mechanism was not used, including the "precondition" option tag in the Require header, the originating UE shall send a new INVITE request using the precondition mechanism, if the originating UE supports the precondition mechanism.
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:	

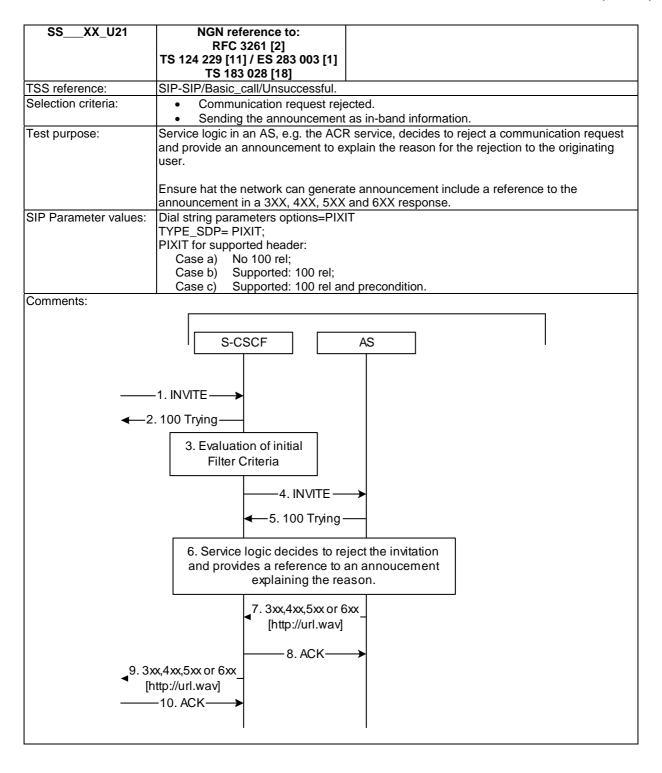
SIXX_U18	PSTN reference to:	NGN reference to:
	EN 300 403-1 [14],	EN 383 001 [15], clause 6.11
	clause G.1.6	ES 283 027 [16], clause 7.2.3.1
TSS reference:	SIP-ISDN/Basic_call/Unsuccessful.	
Selection criteria:		
Test purpose:	During the call establishment, the c	calling user sends a media type not supporte bye the
	SUT. The call attempt is rejected by	ye sending a 415 Unsupported media Type.
SIP Parameter values:	Dial string parameters options=PIX	(IT
	TYPE_SDP= PIXIT;	
	PIXIT for supported header:	
	Case a) No 100 rel;	
	Case b) Supported: 100 rel;	
	Case c) Supported: 100 rel ar	nd precondition
Comments:	SIP	SUT SIP
	INVITE →	→ INVITE
	100 Trying ←	
	415 Unsupported Media Type	 415 Unsupported Media Type
	ACK →	→ ACK

SSXX_U19	NGN reference to: RFC 3261 [2]
	TS 124 229 [11] / ES 283 003 [1] TS 183 028 [18]
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.
Selection criteria:	Communication request rejected.
	Sending the announcement as in-band information.
Test purpose:	Service logic in an AS, e.g. the ACR service, decides to reject a communication request and provide an announcement to explain the reason for the rejection to the originating user. Ensure hat the network can generate announcement using early media i.e. the AS establish an early session and uses that early session to send the in-band announcement.
CID Deremeter velues	
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.



SSXX_U20	NGN reference to:	
	RFC 3261 [2]	
	TS 124 229 [11] / ES 283 003 [1]	
	TS 183 028 [18]	
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.	
Selection criteria:	Communication request rejected.	
	Sending the announcement as in-band information.	
Test purpose:	Service logic in an AS, e.g. the ACR service, decides to reject a communication request	
	and provide an announcement to explain the reason for the rejection to the originating	
	user.	
	Ensure hat the network can generate an anouncement using an established session i.e.	
	the AS accepts the INVITE request and uses the established session to send the in-	
	band announcement.	
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.	





SSXX_U01	NGN reference to:		
	RFC 3261 [2]		
	TS 124 229 [11] / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.		
Selection criteria:			
Test purpose:	Ensure that, when calling to unallocalling user with a 404 Not Found n	,	work initiate call clearing to the
SIP Parameter values:	Dial string parameters options=PIXI TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel ar		
Comments:	SIP	SUT	SIP
	INVITE 100 Trying 404 Not Found ACK	→ ← →	INVITE 404 Not Found ACK

SSXX_U02	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]	
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.	
Selection criteria:		
Test purpose:	Ensure that the call will be released if the clearing to the calling user with a 503 Se	e Service unavailable. The network initiates call ervice unavailable message.
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 presented to the supported of the support of	econdition.
Comments:	SIP INVITE → 100 Trying ← 503 Sevice Unawailable ← ACK →	SUT SIP → INVITE ← 503 Sevice Unawailable → ACK

SSXX_U03	NGN reference to:		
	RFC 3261 [2] and RFC 2327		
	[22]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful		
Selection criteria:			
Test purpose:	Ensure that, when the called user i user with a 486 Busy Here message		cinitiate call clearing to the calling
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel al		
Comments:	SIP INVITE →	SUT	SIP → INVITE
	100 Trying 486 Busy Here ACK →		← 486 Busy Here→ ACK

SSXX_U04	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1	
TSS reference:	SIP-SIP/Basic_call/Unsuccessfu		
Selection criteria:			
Test purpose:	Ensure that when there is no and from INVITE messages), the net 480 Temporarily unavailable me	work initiate call clearing to the c	
SIP Parameter values:	Dial string parameters options=F TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 re Case c) Supported: 100 re	;	
Comments: UE1 INVITE →	Proxy 1	Proxy 2	UE2
	NVITE → INVITE	INVITE INVITE INVITE INVITE INVITE INVITE INVITE INVITE	→ → → → →
	80 ← 480 CK → ACK	← →	

SSXX_U05	NGN reference to:
	draft-ietf-sipping-basic-call-
	flows-02.txt
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.
Selection criteria:	A transaction timer regarding the call establishment is implemented at the UAS.
Test purpose:	Ensure that when there is no answer from the called user, the user initiate call clearing to the calling user with a 480 Temporarily unavailable message (do not disturbe service).
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 UA A INVITE 180 Ringing 480 Teporary unavaible ACK SUT SIP UA B INVITE 180 Ringing 480 Ringing 480 Teporary unavaible ACK

SSXX_U06	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.
Selection criteria:	
Test purpose:	Ensure that when the number is changed, the network initiate call clearing to the calling user with a 410 Gone message.
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.

SSXX_U06	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]			
Comments:	SIP	SUT		SIP
	INVITE	•	→ INVITE	
	100 Trying			
	410m Gone	_	← 410 Gone	
	ACK -	•	→ ACK	

SSXX_U07	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]	
TSS reference:	SIP-SIP/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that when the destination is o calling user with a 502 Bad Gateway	out of order, the network initiate call clearing to the message.
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	
Comments:	SIP INVITE → 100 Trying ← 502 Bad Gateway ACK →	SUT SIP → INVITE ← 502 Bad Gateway → ACK

SSXX_U08	NGN reference to: RFC 3261 [2]		
	TS 124 229 [11] / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.		
Selection criteria:			
Test purpose:	Ensure that the call will be released initiates call clearing to the calling u		
SIP Parameter values:	Dial string parameters options=PIXI TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and		
Comments:	SIP INVITE → 100 Trying ← 484 Address Inconplete ← ACK →	-	SIP INVITE 484 Address Inconplete ACK

SSXX_U09	NGN reference to:		
	RFC 3261 [2]		
	TS 124 229 [11] / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful		
Selection criteria:			
Test purpose:	Ensure that when the called user is not compatible the network initiates call clearing to		
	the calling user with a 503 Service unavailable message.		
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.		

SSXX_U09	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003	[1]			
Comments:	SIP		SUT		SIP
	INVITE	→		→	INVITE
	100 Trying	←			
	503 Service unavailable	←		←	503 Service unavailable
	ACK	→		→	ACK

SSXX_	U10		GN referen RFC 3261 29 [11] / ES					
TSS reference:				Insuccessful.				
Selection criter	ia:							
Test purpose:							NCEL message L message to t	
SIP Parameter	value	S: Dial string TYPE_SI PIXIT for Case Case	g parameter DP= PIXIT; supported l a) No 100 b) Suppo	rs options=PI header:	XIT		·	
Comments:								
Case a)	UE1		Proxy 1		Proxy 2		UE2	
INVITE	→		_		_			
100	←	INVITE	→		_			
CANCEL	→	100 CANCEL	←	INVITE	→	INVITE INVITE INVITE INVITE INVITE INVITE INVITE INVITE CANCEL		
Case b) INVITE 100	→	INVITE 100	→	INVITE	→			
						INVITE INVITE INVITE INVITE INVITE INVITE INVITE		
CANCEL	→	CANCEL	→	CANCEL	→	180 CANCEL	← →	
Case c)						OANOLL	•	
INVITE 100	→	INVITE 100	→	INVITE	→	INVITE INVITE	→	
						INVITE INVITE INVITE INVITE INVITE		
CANCEL	→	CANCEL	→	CANCEL	→	200 OK CANCEL	← →	

SSXX_U11	NGN reference to:					
	RFC 3261 [2]					
	TS 124 229 [11] / ES 283 003 [1]					
TSS reference:	SIP-SIP/Basic_call/Unsuccessful					
Selection criteria:						
Test purpose:	During the session, the calling user decide to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session.					
	Ensutre that if the other party does					
SIP Parameter values:	such as 488 (Not Acceptable Here) Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel ar	ΙΤ	S an ACK.			
Comments:	SIP	SUP	SIP			
	INVITE → 100 Trying ←	-2	INVITE			
	183 Session Progress 180 Ringing 200 OK INVITE ACK RE-INVITE offer 2 488 Not Acceptable Here BYE 200 OK BYE ### ACK ### ACK	Communication Communication	200 OK INVITE ACK RE-INVITE offer 2 488 Not Acceptable Here			

SSXX_U12	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]					
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.					
Selection criteria:						
Test purpose:	During the session, the called user decide to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. Ensutre that if the other party does not accept the change, he sends an error response					
	such as 488 (Not Acceptable Here), which also receives an ACK.					
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 100 Trying 183 Session Progress 180 Ringing 200 OK INVITE ACK RE-INVITE offer 2 488 Not Acceptable Here Tommunication BYE 200 OK BYE RIVITE 183 Session Progress 180 Ringing 200 OK INVITE 400 OK					

SSXX_U13	NGN reference to: RFC 3261 [2]		
	TS 124 229 [11] / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.		
Selection criteria:			
Test purpose:	 Ensure that answer relate message. The media street Ensure that the call is rejuited. 	eam is rejected (port no	·
SIP Parameter values:	Dial string parameters options=PIXTYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel a		
Comments:	SIP	SUT	SIP
	INVITE offer1 →	→	INVITE offer1
	100 Trying ←		
	183 Session Progress ←		183 Session Progress
	180 Ringing answer 1 ←	+	180 Ringing answer 1
	Case A)		
	CANCEL	←	CANCEL
	200 OK →	→	200 OK
	487 Request Terminated ←	←	487 Request Terminated
	ACK →	→	ACK
	Case B)		
	BYE ←		BYE
	200 OK →		200 OK
	487 Request Terminated		487 Request Terminated
	ACK →	→	ACK

SSXX_U14	NGN reference to: RFC 3261 [2] TS 124 229 / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.		
Selection criteria:			
Test purpose:	 Ensure that answer related Progress message. The me Ensure that the call is reject 	dia stream is rejected	(port number is set to zero).
SIP Parameter values:	Dial string parameters options=PIXITYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel ar	nd precondition.	
Comments:	SIP	SUT	SIP
	INVITE offer1 100 Trying 183 Session Progress answer 1	→	INVITE offer1 183 Session Progress answer 1
	Case A)	-	0411051
	CANCEL ←		CANCEL 200 OK
	487 Request Terminated ← ACK →	←	487 Request Terminated ACK
	Case B) BYE		BYE
	200 OK → 487 Request Terminated ←		200 OK
	487 Request Terminated ← ACK →	→	487 Request Terminated ACK

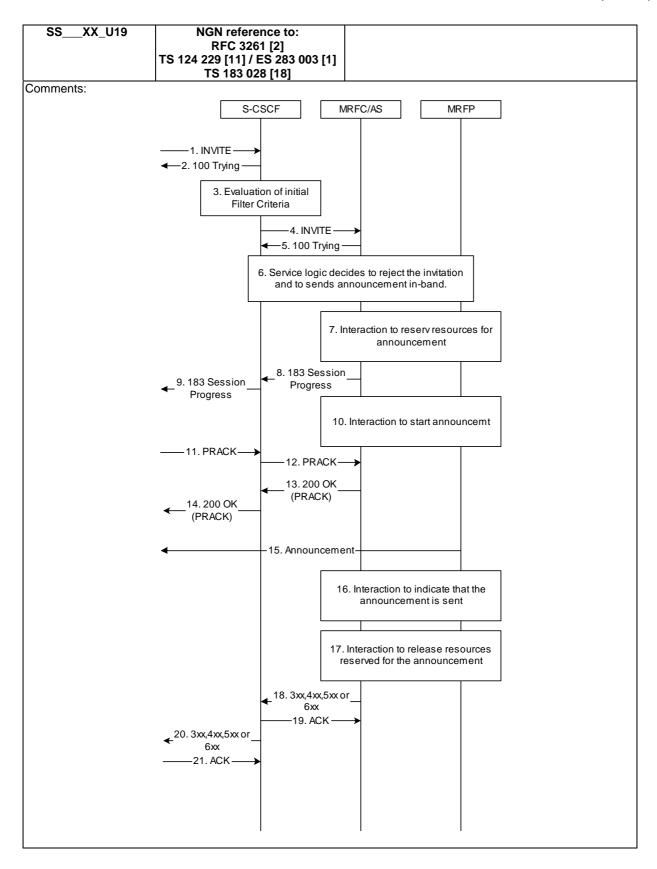
SSXX_U15	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.		
Selection criteria:			
Test purpose:	 Ensure that answer related message. The media strear Ensure that the call is reject 	n is rejected (port r	,
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel ar		
Comments:	SIP INVITE offer1 → 100 Trying ← 200 OK answer 1 → ACK → BYE ← 200 OK →	SUT	SIP → INVITE offer1 → 200 OK answer 1 → ACK ← BYE → 200 OK

SSXX_U16	NGN reference to: RFC 3261 [2]		
	TS 124 229 [11] / ES 283 003 [1]		
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.		
Selection criteria:			
Test purpose:	Ensure that when there is no answe calling user initiate call clearing to the		
SIP Parameter values:	Dial string parameters options=PIXI TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel an		
Comments:	SIP INVITE 100 Trying 180 Ringing CANCEL 200 OK CANCEL 487 Request Terminated ACK SIP 4 4 5 7 8 6 7	SUT →	SIP INVITE 180 Ringing CANCEL 200 OK CANCEL 487 Request Terminated ACK

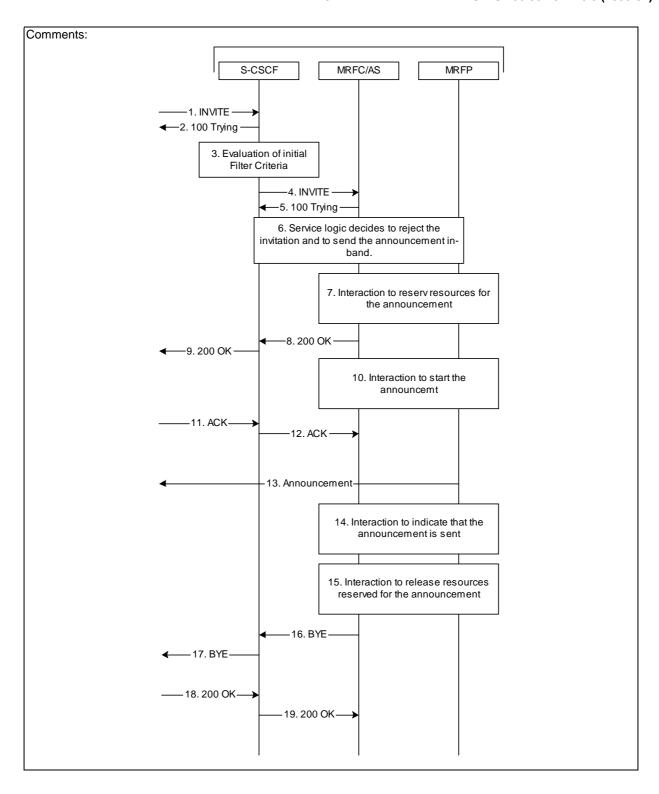
SSXX_U17	NGN reference to: RFC 3261 [2]
	TS 124 229 / ES 283 003 [1]
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.
Selection criteria:	
Test purpose:	Ensure that upon receiving a 421 (Extension Required) response to an initial INVITE request in which the precondition mechanism was not used, including the "precondition" option tag in the Require header, the originating UE shall send a new INVITE request using the precondition mechanism, if the originating UE supports the precondition mechanism.
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:	

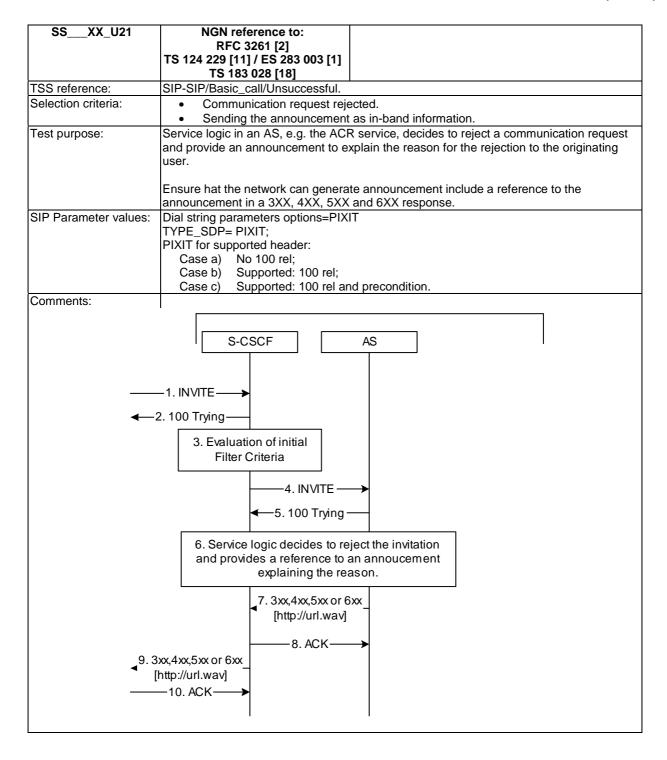
SIXX_U18	PSTN reference to:	NG	N reference to:
	EN 300 403-1 [14],	Q.19	12.5 clause 6.11
	clause G.1.6	EN 383	001 [15] clause 6.11
		ES 283 0	27 [16] clause 7.2.3.1
TSS reference:	SIP-ISDN/Basic_call/Unsuccess	ul.	
Selection criteria:			
Test purpose:	During the call establishment, the	calling user sends a	media type not supporte bye the
	SUT. The call attempt is rejected	bye sending a 415 Uns	supported media Type.
SIP Parameter values:	Dial string parameters options=F	IXIT	
	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
	100 Trying	(
	415 Unsupported Media Type	← ←	415 Unsupported Media Type
	ACK	→	ACK

SSXX_U19	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1] TS 183 028 [18]
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.
Selection criteria:	Communication request rejected.Sending the announcement as in-band information.
Test purpose:	Service logic in an AS, e.g. the ACR service, decides to reject a communication request and provide an announcement to explain the reason for the rejection to the originating user. Ensure hat the network can generate announcement using early media i.e. the AS establish an early session and uses that early session to send the in-band announcement.
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.



SSXX_U20	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1] TS 183 028 [18]
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.
Selection criteria:	Communication request rejected.Sending the announcement as in-band information.
Test purpose:	Service logic in an AS, e.g. the ACR service, decides to reject a communication request and provide an announcement to explain the reason for the rejection to the originating user. Ensure hat the network can generate an anouncement using an established session i.e. the AS accepts the INVITE request and uses the established session to send the inband announcement.
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.





6.5.2 Test purposes for SIP-SIP, Supplementary services

6.5.2.1 Test purposes for OIP

SSXXSS_OIP01	OIP/OIR reference [16] clauses 4	1.3.2 and 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/OIP			
Selection criteria:	The user subscribes OIR "temporar		ot restricted".	
	The terminating user subscribes the			
Test purpose:	In case "temporary mode" with default "not restricted" is subscribed and no privacy value is received, ensure that no identity information is provided by the originating UE, the terminating user receives a P-Asserted-Identity based on the default public user			
OID D	identity associated with the originati		ie originator or the session.	
SIP Parameter values:	Dial string parameters options=PIXI TYPE_SDP= PIXIT;	ı		
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;			
	Case c) Supported: 100 rel an	d precondition.		
Comments:	SIP UA A	SUT	SIP UA B	
	INVITE →		→ INVITE	
	180 Ringing ←		← 180 Ringing	
	200 OK INVITE ←		← 200 OK INVITE	
	ACK → ACK			
	BYE →	Conversation	→ BYE	
	200 OK (BYE)		← 200 OK (BYE)	

SSXXSS_OIP02	OIP/OIR reference [19] clauses				
TSS reference:	SIP-SIP/SupplementaryServices/OIP.				
Selection criteria:	The user subscribes OIR "tempora	ry mode" default "i	not restricted".		
	The terminating user subscribes th	e OIP service.			
Test purpose:	In case "temporary mode" with def	ault "not restricted	is subscribed and the Privacy		
	header field that is set to "none", e				
	originating UE, the terminating use	r receives a P-Ass	serted-Identity based on the default		
	public user identity associated with	the originating UE	identifies the originator of the		
	session.				
SIP Parameter values:	Dial string parameters options=PIX	ΊΤ			
	TYPE_SDP= PIXIT;				
	PIXIT for supported header:				
	Case a) No 100 rel;				
		Case b) Supported: 100 rel;			
	Case c) Supported: 100 rel a				
Comments:	SIP UA A	SUT	SIP UA B		
	INVITE →		→ INVITE		
	180 Ringing ←		€ 180 Ringing		
	200 OK INVITE ←		€ 200 OK INVITE		
	ACK → ACK				
	DV5	Conversation	3 DVE		
	BYE ->		→ BYE		
	200 OK (BYE) ←		← 200 OK (BYE)		
		_			

SSXXSS_OIP03	OIP/OIR reference [19] clauses	4.3.2 and 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.			
Selection criteria:	The user subscribes OIR "tempora	ary mode" default "r	estricted".	
	The terminating user subscribes the	ne OIP service.		
Test purpose:	In case "temporary mode" with de field that is set to " none ", ensure t		ubscribed and the Privacy header	
			erted-Identity based on the default	
	public user identity associated with			
	session.	Title originating OL	identified the originator of the	
SIP Parameter values:	Dial string parameters options=PI	XIT		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;			
	Case c) Supported: 100 rel a			
Comments:	SIP UA A	SUT	SIP UA B	
	INVITE ->		→ INVITE	
	180 Ringing ←		← 180 Ringing	
	200 OK INVITE		← 200 OK INVITE	
	ACK ->		→ ACK	
	5.45	Conversation	> 5.45	
	BYE -		→ BYE	
	200 OK (BYE) ←	•	← 200 OK (BYE)	

SSXXSS_OIP05	OIP/OIR reference [19]		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.			
Selection criteria:	The user subscribes OIR "temporal	ry mode" default "n	ot restricted".	
	The terminating user subscribes the	e OIP service.		
Test purpose:	In case "temporary mode" with defa			
	is received, ensure that an identity			
	not match with the set of registered			
	user receives a P-Asserted-Identit			
	with the originating UE identifies the		ession.	
SIP Parameter values:	Dial string parameters options=PIX	IT		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;			
	Case c) Supported: 100 rel ar			
Comments:	SIP UA A	SUT	SIP UA B	
	INVITE →		→ INVITE	
	180 Ringing ←		← 180 Ringing	
	200 OK INVITE		← 200 OK INVITE	
	ACK →		→ ACK	
		Conversation		
	BYE →		→ BYE	
	200 OK (BYE) ←		← 200 OK (BYE)	

SSXXSS_OIP06	OIP/OIR reference [19]			
TSS reference:	SIP-SIP/SupplementaryServices/OIP.			
Selection criteria:	The user subscribes OIR "temporary mode" default "not restricted".			
	The terminating user subscribes the OIP service.			
Test purpose:	In case "temporary mode" with default "not restricted" is subscribed and the Privacy header field that is set to " none ", ensure that an identity information is provided by the originating UE, 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 identifies the originator of the session.			
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 UA A SUT SIP UA B INVITE → INVITE 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → ACK Conversation BYE → BYE 200 OK (BYE) ← 200 OK (BYE)			

SSXXSS_OIP07	OIP/OIR reference [19] clauses 4	.3.2 and 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.			
Selection criteria:	The user subscribes OIR "temporary	y mode" default "r	estricted".	
	The terminating user subscribes the	OIP service.		
Test purpose:	In case "temporary mode" with default "restricted" is subscribed and the Privacy header field that is set to " none ", ensure that an identity information is provided by the originating UE, 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 identifies the originator of the session.			
SIP Parameter values:	Dial string parameters options=PIXI TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and			
Comments:	SIP UA A INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)	SUT Conversation	SIP UA B → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK (BYE)	

SSXXSS_OIP09	OIP/OIR reference [19] clauses	4.3.2 and 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.			
Selection criteria:	The user subscribes OIR "tempora	ry mode" default "n	ot restricted".	
	The terminating user subscribes th	e OIP service.		
Test purpose:	In case "temporary mode" with def	ault "not restricted"	is subscribed and no privacy value	
	is received, ensure that an identity			
	matches with the set of registered			
	user receives a P-Asserted-Identi		ormation provided by the	
	originating UE identifies the origina			
SIP Parameter values:	Dial string parameters options=PIX	IT		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;			
	Case c) Supported: 100 rel a			
Comments:	SIP UA A	SUT	SIP UA B	
	INVITE →		→ INVITE	
	180 Ringing ←		← 180 Ringing	
	200 OK INVITE		€ 200 OK INVITE	
	ACK →		→ ACK	
	5)/5	Conversation	3 DVE	
	BYE ->		→ BYE	
	200 OK (BYE) ←		← 200 OK (BYE)	

SSXXSS_OIP10	OIP/OIR reference [19] clauses	4.3.2 and 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.			
Selection criteria:	The user subscribes OIR "tempora	ry mode" default "r	not restricted".	
	The terminating user subscribes th	e OIP service.		
Test purpose:	In case "temporary mode" with def	ault "not restricted"	is subscribed and the Privacy	
	header field that is set to "none", e	nsure that an ident	tity information is provided by the	
	originating UE, matches with the se			
	the terminating user receives a P-A	Asserted-Identity	based on the information provided	
	by the originating UE, identifies the	originator of the s	ession.	
SIP Parameter values:	Dial string parameters options=PIX	IT		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;			
	Case c) Supported: 100 rel a			
Comments:	SIP UA A	SUT	SIP UA B	
	INVITE →		→ INVITE	
	180 Ringing ←		← 180 Ringing	
	200 OK INVITE		← 200 OK INVITE	
	ACK →		→ ACK	
		Conversation		
	BYE →		→ BYE	
	200 OK (BYE) ←		← 200 OK (BYE)	

SSXXSS_OIP11	OIP/OIR reference [19] clauses	4.3.2 and 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.			
Selection criteria:	The user subscribes OIR "tempora	ry mode" default "r	estricted".	
	The terminating user subscribes the OIP service.			
Test purpose:	In case "temporary mode" with def field that is set to " none ", ensure the	nat an identity infor	mation is provided by the	
	originating UE, matches with the sithe terminating user receives a P- <i>I</i> by the originating UE, identifies the	Asserted-Identity	pased on the information provided	
SIP Parameter values:			ession.	
SIP Parameter values.	Dial string parameters options=PI	.11		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;			
	Case c) Supported: 100 rel a	nd precondition.		
Comments:	SIP UA A	SUT	SIP UA B	
	INVITE →		→ INVITE	
	180 Ringing ←		← 180 Ringing	
	200 OK ÎNVITE ←			
	ACK → ACK			
	Conversation			
	BYE →	Convolation	→ BYE	
	200 OK (BYE)			
	200 011 (012)			

SSXXSS_OIP13	OIP/OIR reference [19]	clause 4.5.2.3		
TSS reference:	SIP-SIP/SupplementaryServices/OIP.			
Selection criteria:	The S-CSCF has knowledge of	The S-CSCF has knowledge of an associated tel-URI for a SIP URI		
	The terminating user subscribe	s the OIP service.		
Test purpose:	The S-CSCF has knowledge of	an associated tel-URI	for a SIP URI contained in the P-	
			the S-CSCF adds a second P-	
SIP Parameter values:	Asserted-Identity header field containing this tel-URI. 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 UA A	SUT	SIP UA B	
Comments.	INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)	Conversation	 → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK (BYE) 	

SSXXSS_OIP14	OIP/OIR reference [19] clau	se 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/OIP			
Selection criteria:	OIP service subscribed.			
	Special arrangement does not exist.	Special arrangement does not exist.		
	The network attempts to match the information in the From header with the set of			
	registered public identities of the original	ginating user.		
Test purpose:	The special arrangement does not e			
	in the From header with the set of re			
	There is no match found, the AS set		to the SIP URI that includes the	
	registered default public user identity			
SIP Parameter values:	Dial string parameters options=PIXI	Т		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;			
	Case c) Supported: 100 rel and			
Comments:	SIP UA A	SUT	SIP UA B	
	INVITE →		→ INVITE	
	180 Ringing ←		← 180 Ringing	
	200 OK INVITE ←		← 200 OK INVITE	
	ACK →		→ ACK	
	DV5	Conversation	> DV5	
	BYE -		→ BYE	
	200 OK (BYE) ←		← 200 OK (BYE)	

SSXXSS_OIP15	OIP/OIR reference [19] c	ause 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices	OIP.	
Selection criteria:	OIP service subscribed.		
	Special arrangement exists.		
Test purpose:	The special arrangement exists. The network does not attempt to match the information in the From header with the set of registered public identities of the originating user. The From header field is transparently transported to the terminating user.		
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 UA A	SUT	SIP UA B
	180 Ringing 200 OK INVITE ACK BYE	Conversation	→ INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE
	200 OK (BYE)	7	← 200 OK (BYE)

6.5.2.2 Test purposes for OIR

SSXXSS_OIR01	OIP/OIR reference [19] clause	es 4.3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The user subscribes OIR "tempo	rary mode" default "r	estricted".
Test purpose:	In case "temporary mode" with default "restricted" is subscribed a Privacy value was not received, ensure that no identity information is provided by the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
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 UA A	SUT	SIP UA B
	200 OK INVITE ACK	Conversation	→ INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK (BYE)

SSXXSS_OIR02	OIP/OIR reference [19] clauses 4.3.2 and 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The user subscribes OIR "temporary mode" default "restricted".		
Test purpose:	In case "temporary mode" with default "restricted" is subscribed a Privacy header field that is set to "id", ensure that no identity information is provided by the originating UE, the terminating user does not receive an information identifying the originator of the session.		
	The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
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 UA A SUT SIP UA B		
	INVITE		

SSXXSS_OIR03	OIP/OIR reference [19] clauses 4	.3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The user subscribes OIR "temporary	/ mode" default "no	t restricted".
Test purpose:	In case "temporary mode" with default "not restricted" is subscribed a Privacy header field that is set to "id", ensure that no identity information is provided by the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
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 UA A INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)	SUT -	SIP UA B NVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)

SSXXSS_OIR04	OIP/OIR reference [19] clauses 4.3.2 and 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The user subscribes OIR permanent mode.		
Test purpose:	In case "permanent mode" is subscribed, a Privacy value was not received, ensure that no identity information is provided by the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
SIP Parameter values:	TYPE_SDP= PIXIT; Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel; precondition.		
Comments:	SIP UA A SUT SIP UA B INVITE → → INVITE 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → ACK Conversation → BYE 200 OK (BYE) ← 200 OK (BYE)		

SSXXSS_OIR05	OIP/OIR reference [19] clause	s 4.3.2 and 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The user subscribes OIR permar	nent mode.		
Test purpose:	In case "permanent mode" is subscribed a Privacy header field that is set to "id", ensure that no identity information is provided by the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-			
SIP Parameter values:	Dial string parameters options=P TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel Case c) Supported: 100 rel	;		
Comments:	SIP UA A	SUT	SIP UA B	
	INVITE -	→	→ INVITE	
	180 Ringing	-	← 180 Ringing	
	200 OK INVITE	-	← 200 OK INVITE	
	ACK -	→	→ ACK	
		Conversation		
	BYE -	→	→ BYE	
	200 OK (BYE)	E	← 200 OK (BYE)	

SSXXSS_OIR06	OIP/OIR reference [19] clauses	4.3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The user subscribes OIR "tempora	ary mode" default "re	estricted".
Test purpose:	In case "temporary mode" with default "restricted" is subscribed a Privacy value was not received, ensure that an identity information is provided by the originating UE, does not match with the set of registered public identities of the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
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 UA A INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE) SIP UA A → ACK → ACK → ACK → ACK → ACK → ACK	SUT	SIP UA B INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)

SSXXSS_OIR07	OIP/OIR reference [19] clauses	4.3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The user subscribes OIR "tempora	ry mode" default "re	estricted".
Test purpose:	In case "temporary mode" with default "restricted" is subscribed a Privacy header field that is set to "id", ensure that an identity information is provided by the originating UE, does not match with the set of registered public identities of the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
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 UA A	SUT	SIP UA B
	INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE) →	Conversation	→ INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK (BYE)

SSXXSS_OIR08	OIP/OIR reference [19] clauses	4.3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR		
Selection criteria:	The user subscribes OIR "tempora	ry mode" default "r	ot restricted".
Test purpose:	In case "temporary mode" with default "not restricted" is subscribed a Privacy header field that is set to "id", ensure that an identity information is provided by the originating UE, does not match with the set of registered public identities of the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
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 UA A INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)	SUT Conversation	SIP UA B → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK (BYE)

SSXXSS_OIR09	OIP/OIR reference [19] clauses 4.	3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR		
Selection criteria:	The user subscribes OIR permanent	mode.	
Test purpose:	In case "permanent mode" is subscribed, a Privacy value was not received, ensure that an identity information is provided by the originating UE, does not match with the set of registered public identities of the originating UE, the terminating user does not receive an information identifying the originator of the session- The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
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		
Comments:	SIP UA A INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE) SIP UA A → 180 Ringing ← 200 OK (NVITE ← 200 OK (BYE)	SUT Conversation	SIP UA B NVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)

SSXXSS_OIR10	OIP/OIR reference [19] clauses	4.3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The user subscribes OIR permaner	nt mode.	
Test purpose:	In case "permanent mode" is subscribed a Privacy header field that is set to "id", ensure that an identity information is provided by the originating UE, does not match with the set of registered public identities of the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel ar	IT	
Comments:	SIP UA A INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE) SIP UA A	SUT Conversation	SIP UA B INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)

SSXXSS_OIR11	OIP/OIR reference [19] clauses	4.3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR		
Selection criteria:	The user subscribes OIR "tempora	ry mode" default "re	estricted".
Test purpose:	In case "temporary mode" with default "restricted" is subscribed a Privacy value was not received, ensure that an identity information is provided by the originating UE, matches with the set of registered public identities of the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel ar		
Comments:	SIP UA A INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE) SIP UA A → HOTELLINE ACK → HOTELLINE ACK → HOTELLINE ACK → HOTELLINE ACK → HOTELLINE ACK HOTELLINE HOTELLINE	SUT	SIP UA B INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)

SSXXSS_OIR12	OIP/OIR reference [19] clauses	4.3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR		
Selection criteria:	The user subscribes OIR "tempora	ary mode" default "r	estricted".
Test purpose:	In case "temporary mode" with default "restricted" is subscribed a Privacy header field that is set to "id", ensure that an identity information is provided by the originating UE, matches with the set of registered public identities of the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
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 UA A INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE) SIP UA A → ACK →	SUT Conversation	SIP UA B → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK (BYE)

SSXXSS_OIR13	OIP/OIR reference [19] clauses 4.	3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR		
Selection criteria:	The user subscribes OIR "temporary	mode" default "no	ot restricted".
Test purpose:	a Privacy header field that is set to "id", ensure that an identity information is provided by the originating UE, matches with the set of registered public identities of the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
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		
Comments:	SIP UA A INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)	SUT Conversation	SIP UA B → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK (BYE)

SSXXSS_OIR14	OIP/OIR reference [19] clauses	4.3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR		
Selection criteria:	The user subscribes OIR permane	nt mode.	
Test purpose:	In case "permanent mode" is subscribed, a Privacy value was not received, ensure that an identity information is provided by the originating UE, matches with the set of registered public identities of the originating UE, the terminating user does not receive an information identifying the originator of the session- The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel a	ΊΤ	
Comments:	SIP UA A INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)	SUT Conversation	SIP UA B → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK (BYE)

SSXXSS_OIR15	OIP/OIR reference [19] clauses 4.3	3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR		
Selection criteria:	The user subscribes OIR permanent r	mode.	
Test purpose:	In case "permanent mode" is subscribed a Privacy header field that is set to "id", ensure that an identity information is provided by the originating UE, matches with the set of registered public identities of the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
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		
Comments:	SIP UA A INVITE → 180 Ringing ← 200 OK INVITE ← ACK →	SUT Conversation	SIP UA B → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK (BYE)

SSXXSS_OIR16	OIP/OIR reference [19] c	ause 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The user subscribes OIR perman	ent mode.		
	Modifies the From header field to remove the identification.			
Test purpose:	In case of the OIR service, the AS	6 modifies the From I	header field to remove the	
			a INVITE message where the he	
	SIP From header field is set to "a	nonymous" or "unava	ailable" and no P-Asserted-Identity	
	header is received-			
SIP Parameter values:	Dial string parameters options=P	XIT		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 10;0 rel			
	Case c) Supported: 100 rel	•		
Comments:	SIP UA A	SUT	SIP UA B	
	INVITE		→ INVITE	
	180 Ringing		← 180 Ringing	
	200 OK INVITE		← 200 OK INVITE	
	ACK =		→ ACK	
		Conversation		
	BYE		→ BYE	
	200 OK (BYE)	•	← 200 OK (BYE)	

SSXXSS_OIR17	OIP/OIR reference [19] claus	ses 4.3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIR.		
Selection criteria:	The user subscribes OIR perm the AS add a Privacy header fi		
Test purpose:	In case of the OIR service, the		der field set to "user".
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 UA A INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK (BYE)	SUT Conversation	SIP UA B → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK (BYE)

SSXXSS_OIR18	OIP/OIR reference [19] clauses 4.3.2 and 4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/OIP		
Selection criteria:	The user subscribes OIR "permanent mode".		
	The terminating user subscribes the OIP service.		
Test purpose:	In case "permanent mode" is subscribed a Privacy header field that is set to "none", ensure that an identity information is provided by the originating UE, matches with the set of registered public identities of the originating UE, the terminating user does not receive an information identifying the originator of the session. The terminating UE receives a INVITE message where the he SIP From header field is set to "anonymous" or "unavailable" and no P-Asserted-Identity header is received-		
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 UA A SUT SIP UA B INVITE → INVITE 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → ACK Conversation BYE → BYE 200 OK (BYE) ← 200 OK (BYE)		

SSXXSS_OIR19	OIP/OIR reference [19] clauses	s 4.3.2 and4.5.2.4		
TSS reference:	SIP-SIP/SupplementaryServices/	SIP-SIP/SupplementaryServices/OIP.		
Selection criteria:	The user subscribes OIR "permanent mode".			
	The terminating user subscribes t	he OIP service.		
Test purpose:	In case "permanent mode" is subs	scribed a Privacy he	ader field that is set to " none ",	
			originating UE, does not match with	
			UE, the terminating user does not	
	receive an information identifying			
	receives a INVITE message wher			
	or "unavailable" and no P-Asserte		received-	
SIP Parameter values:	Dial string parameters options=PL	XIT		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;			
	Case c) Supported: 100 rel a			
Comments:	SIP UA A	SUT	SIP UA B	
	INVITE		→ INVITE	
	180 Ringing		← 180 Ringing	
	200 OK INVITE		€ 200 OK INVITE	
	ACK -		→ ACK	
	BYE -	Conversation	→ BYE	
	200 OK (BYE) ←	_	← 200 OK (BYE)	

SSXXSS_OIR20	OIP/OIR reference [19] clause	es 4.3.2 and 4.5.2.4	
TSS reference:	SIP-SIP/SupplementaryServices/OIP.		
Selection criteria:	The user subscribes OIR "perma	anent mode".	
	The terminating user subscribes	the OIP service.	
Test purpose:	In case "permanent mode" is su	bscribed a Privacy he	eader field that is set to " none ",
	ensure that no identity information	on is provided by the	originating UE, does not match with
	the set of registered public ident	ities of the originating	UE, the terminating user does not
	receive an information identifying	g the originator of the	session. The terminating UE
	receives a INVITE message who	ere the he SIP From h	neader field is set to "anonymous"
	or "unavailable" and no P-Asser	ted-Identity header is	received.
SIP Parameter values:	Dial string parameters options=F	PIXIT	
	TYPE_SDP= PIXIT;		
	PIXIT for supported header:		
	Case a) No 100 rel;		
	Case b) Supported: 100 re		
	Case c) Supported: 100 re	l and precondition.	
Comments:	SIP UA A	SUT	SIP UA B
	INVITE	→	→ INVITE
	180 Ringing	←	← 180 Ringing
		←	← 200 OK INVITE
	ACK	→	→ ACK
		Conversation	
		→	→ BYE
	200 OK (BYE)	(← 200 OK (BYE)

6.5.2.3 Test purposes for TIP

SSXXSS_TIP01	[18] clause 5.2.6.4, [12] clause 4.5.2.	.4		
TSS reference:	SIP-SIP/SupplementaryServices/TIP.			
Selection criteria:	The user subscribes TIR "temporary mode" default "not restricted".			
	The originating user subscribes the TIP service.			
Test purpose:	In case "temporary mode" with default "not restricted" is subscribed and no privacy value			
	is received, ensure that the identity information is provided by the P-CSCF, the			
	originating user receives a P-Asserted-Identity based on the default public user identity			
OID D	associated with the terminating UE identifies the callee.			
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 precond	dition		
Comments:	SIP UA A		UT	SIP UA B
	INVITE	→	→	INVITE
	Case A			
	180 Ringing (P-Asserted-Identiy)	←	←	180 Ringing
	200 OK INVITE	←	←	200 OK INVITE
	ACK	→	→	ACK
	Case B	_	_	
	183 Session Progress (P-Asserted-Identiy)	(183 Session Progress
	180 Ringing	(180 Ringing
	200 OK INVITE	(200 OK INVITE
	ACK	→	7	ACK
	Case C	←	_	190 Binging
	180 Ringing 200 OK INVITE (P-Asserted-Identiy)	-		180 Ringing 200 OK INVITE
	ACK	→		ACK
	Conversation	•	•	7.010
	BYF	→	→	BYF
	200 OK (BYE)	÷	_	200 OK (BYE)

SSXXSS_TIP02	[18] clause 5.2.6.4, [12] clause 4.5.2	.4				
TSS reference:	SIP-SIP/SupplementaryServices/TIP.					
Selection criteria:	The user subscribes TIR "temporary mode" default "not restricted".					
	The originating user subscribes the TIP servi					
Test purpose:	In case "temporary mode" with default "not re					
	header field is set to "none", ensure that the					
	CSCF, the originating user receives a P-Ass					
	user identity associated with the terminating	UE identif	ies the c	allee.		
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 precon	dition				
Comments:	SIP UA A	SL	IT	SIP UA B		
Comments.	INVITE	→ 3C	,, →	INVITE		
	Case A	7	7	INVITE		
	180 Ringing(P-Asserted-Identiy)	←	(180 Ringing		
	200 OK INVITE	È	È			
	ACK	→	→	ACK		
	Case B			_		
	183 Session Progress(P-Asserted-Identiy)	←	←	183 Session Progress		
	180 Ringing	←	←	180 Ringing		
	200 OK INVITE	←	←	200 OK INVITE		
	ACK	→	→	ACK		
	Case C					
	180 Ringing ← 180 Ringing					
	200 OK INVITE(P-Asserted-Identiy) ← 200 OK INVITE					
	ACK → ACK					
	Conversation	_	_			
	BYE	→		BYE		
	200 OK (BYE)	←	+	200 OK (BYE)		

SSXXSS_TIP03	[18] clause 5.2.6.4, [12] clause 4.5.2	.4				
TSS reference:	SIP-SIP/SupplementaryServices/TIP.					
Selection criteria:	The user subscribes TIR "temporary mode" default "restricted".					
	The originating user subscribes the TIP servi					
Test purpose:	In case "temporary mode" with default "restriction					
	field is set to "none" ensure that the identity i					
	P-CSCF, the originating user receives a P-As					
	user identity associated with the terminating	UE identif	ies the c	allee.		
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;	dition				
Comments:	Case c) Supported: 100 rel and precond	SU	IT	SIP UA B		
Comments.	INVITE	→ SU	′¹ →	INVITE		
	Case A	7	7	INVITE		
	180 Ringing(P-Asserted-Identiy)	←	←	180 Ringing		
	200 OK INVITE	È	è	200 OK INVITE		
	ACK	À	À	ACK		
	Case B	-	-	7.6.1		
	183 Session Progress(P-Asserted-Identiy)	←	←	183 Session Progress		
	180 Ringing	←	←			
	200 OK INVITE	←	←	200 OK INVITE		
	ACK	→	→	ACK		
	Case C					
	180 Ringing ← 180 Ringing					
	200 OK INVITE(P-Asserted-Identiy) ← ← 200 OK INVITE					
	ACK → ACK					
	Conversation					
	BYE	→		BYE		
	200 OK (BYE)	←	+	200 OK (BYE)		
				.,		

6.5.2.4 Test purposes for TIR

SSXXSS_TIR01	[18] clause 5.2.6.4, [12] clauses 4.5.2.4	and 4.5.2.9				
TSS reference:	SIP-SIP/SupplementaryServices/TIR.					
Selection criteria:	The terminating user subscribes TIR "temporary mode" default "not restricted".					
	The originating user subscribes the TIP ser	vice.				
Test purpose:	In case "temporary mode" with default "not					
	header field is set to "id", ensure that identi					
	P-CSCF, the originating user does not rece		ion identifying the callee			
OID D	neither in a 1XXx response nor in a 2XX re	sponse.				
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;					
0	Case c) Supported: 100 rel and preco		OID HA D			
Comments:	SIP UA A	SUT	SIP UA B			
	INVITE	→	→ INVITE			
	Case A	_	400 Dinging			
	180 Ringing 200 OK INVITE	(← 180 Ringing ← 200 OK INVITE			
		→				
	ACK Case B	7	→ ACK			
		_	4.00 Cassian Drawnson			
	183 Session Progress	(← 183 Session Progress			
	180 Ringing	(← 180 Ringing			
	200 OK INVITE ← 200 OK INVITE					
	ACK → ACK					
	Conversation	_	→ DVE			
	BYE	→	→ BYE			
	200 OK (BYE)	←	← 200 OK (BYE)			
		·				

SSXXSS_TIR02	[18] clause 5.2.6.4, [12] clauses 4.5.2.4 a	nd 4.5.2.9				
TSS reference:	SIP-SIP/SupplementaryServices/TIR.					
Selection criteria:	The terminating user subscribes TIR "temporary mode" default "restricted".					
	The originating user subscribes the TIP serv					
Test purpose:	In case "temporary mode" with default "restricted" is subscribed and no privacy value is received, ensure that identity information is provided by the terminating P-CSCF, the originating user does not receive an information identifying the callee neither in a 1XXx response nor in a 2XX response.					
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 UA A	SUT	SIP UA B			
	INVITE Case A	→	→ INVITE			
	180 Ringing	←	← 180 Ringing			
	200 OK INVITE	(← 200 OK INVITE			
	ACK Case B	→	→ ACK			
	183 Session Progress	←	← 183 Session Progress			
	180 Ringing	←	← 180 Ringing			
	200 OK INVITE	←	← 200 OK INVITE			
	ACK → → ACK Conversation					
	BYE	→	→ BYE			
	200 OK (BYE)	←	← 200 OK (BYE)			

SSXXSS_TIR03	[18] clause 5.2.6.4, [12] clauses 4.5.2.4 a	and 4.5.2.9				
TSS reference:	SIP-SIP/SupplementaryServices/TIR.					
Selection criteria:	The terminating user subscribes TIR "temporary mode" default "restricted".					
	The originating user subscribes the TIP ser	vice.				
Test purpose:	In case "temporary mode" with default "rest					
	field is set to "id", ensure that identity inforn					
	the originating user does not receive an info	ormation identi	ifying	the callee neither in a		
	1XXx response nor in a 2XX response.					
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 preco					
Comments:	SIP UA A	SUT		SIP UA B		
	INVITE	→	→	INVITE		
	Case A	_	_			
	180 Ringing	(180 Ringing		
	200 OK INVITE	(200 OK INVITE		
	ACK _	→	→	ACK		
	Case B	-	_	100 0		
	183 Session Progress	(183 Session Progress		
	180 Ringing	(180 Ringing		
	200 OK INVITE	(200 OK INVITE		
	ACK → ACK					
	Conversation	_	_			
	BYE	→		BYE		
	200 OK (BYE)	←	←	200 OK (BYE)		

SSXXSS_TIR04	[18] clause 5.2.6.4, [12] clauses 4.5.2.4 an	d 4.5.2.9			
TSS reference:	SIP-SIP/SupplementaryServices/TIR.				
Selection criteria:	The terminating user subscribes TIR "permanent mode".				
	The originating user subscribes the TIP service	ce.			
Test purpose:	In case "permanent mode" is subscribed and no privacy value is received, ensure that identity information is provided by the terminating P-CSCF, the originating user does not receive an information identifying the callee neither in a 1XXx response nor in a 2XX response.				
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 nd precondition.				
Comments:	SIP UA A	SUT	SIP UA B		
	INVITE Case A	→	→ INVITE		
	180 Ringing	←	← 180 Ringing		
	200 OK INVITE	←	← 200 OK INVITE		
	ACK Case B	→	→ ACK		
	183 Session Progress	←	← 183 Session Progress		
	180 Ringing	←	← 180 Ringing		
	200 OK INVITE	←	← 200 OK INVITE		
	ACK Conversation	→	→ ACK		
	BYE	→	→ BYE		
	200 OK (BYE)	←	← 200 OK (BYE)		

SSXXSS_TIR05	[18] clause 5.2.6.4, [12] clauses 4.5.2.4	and 4.5.2.9			
TSS reference:	SIP-SIP/SupplementaryServices/TIR.				
Selection criteria:	The terminating user subscribes TIR "permanent mode".				
	The originating user subscribes the TIP se	rvice.			
Test purpose:	In case "permanent mode" is subscribed a				
	that identity information is provided by the				
	not receive an information identifying the ca	allee neither in	a 1XXx response nor in a 2XX		
	response.				
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 preco				
Comments:	SIP UA A	SUT			
	INVITE	→	→ INVITE		
	Case A		400 Diamina		
	180 Ringing	-	← 180 Ringing		
	200 OK INVITE ACK	← →	← 200 OK INVITE→ ACK		
	Case B	7	7 ACK		
	183 Session Progress	←	← 183 Session Progress		
	180 Ringing	(← 180 Ringing		
	200 OK INVITE	-	€ 200 OK INVITE		
	ACK	À	→ ACK		
	Conversation	-	2 /1011		
	BYE	→	→ BYE		
	200 OK (BYE)	÷	€ 200 OK (BYE)		
	,		()		

SSXXSS_TIR16	[18] clause 5.2.6.4, [12] clause 4.3.	3					
TSS reference:	SIP-SIP/SupplementaryServices/TIP						
Selection criteria:	The originating user subscribes the TIP servi						
		The originating use has the "override category".					
	The terminating user subscribes TIR "perma	nent mode	e".				
Test purpose:	Ensure that an originating user, that subscrib override category, receives the P-Asserted-lu- response or in a 2XX response. The originati "id".	dentity ide	entifying t	he callee in a 1XX			
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 precon	dition.					
Comments:	SIP UA A	S	UT	SIP UA B			
	INVITE	→	→	INVITE			
	Case A						
	180 Ringing(P-Asserted-Identiy)	←		180 Ringing			
	200 OK INVITE	←		200 OK INVITE			
	ACK	→	→	ACK			
	Case B	_	_				
	183 Session Progress(P-Asserted-Identiy)	(183 Session Progress			
	180 Ringing	(180 Ringing			
	200 OK INVITE	(200 OK INVITE			
	ACK	→	→	ACK			
	Case C		-	100 B: :			
	180 Ringing ← ★ 180 Ringing						
	200 OK INVITE(P-Asserted-Identiy) ← 200 OK INVITE						
	ACK → ACK						
	Conversation		_	DV.			
	BYE	→	_	BYE			
	200 OK (BYE)	←	←	200 OK (BYE)			

SSXXSS_TIR17	[18] clause 5.2.6.4, [12] clause 4.5.2	.4				
TSS reference:	SIP-SIP/SupplementaryServices/TIP					
Selection criteria:	The user subscribes TIR "permanent mode".					
	The originating user subscribes the TIP servi	ce.				
Test purpose:	In case "permanent mode" is subscribed and					
	ensure that the identity information is provide					
	receives a P-Asserted-Identity based on the	e default p	oublic us	er identity associated with		
	the terminating UE identifies the callee.					
SIP Parameter values:	Dial string parameters options=PIXIT					
	TYPE_SDP= PIXIT;					
	PIXIT for supported header: Case a) No 100 rel;					
	Case a) No 100 rel; Case b) Supported: 100 rel;					
	Case b) Supported: 100 rel and precond	dition				
Comments:	SIP UA A	SU	Т	SIP UA B		
Comments.	INVITE	→		INVITE		
	Case A	-	-	1144112		
	180 Ringing(P-Asserted-Identiy)	←	←	180 Ringing		
	200 OK INVITE	←		200 OK INVITE		
	ACK	→	→	ACK		
	Case B					
	183 Session Progress(P-Asserted-Identiy)	←		183 Session Progress		
	180 Ringing	←		180 Ringing		
	200 OK INVITE	←		200 OK INVITE		
	ACK	→	→	ACK		
	Case C	_	_	100 B: :		
	180 Ringing ← 180 Ringing					
	200 OK INVITE(P-Asserted-Identiy) 200 OK INVITE					
	ACK → ACK Conversation					
	BYE	→	-	BYE		
	200 OK (BYE)	÷		200 OK (BYE)		
	(,			(/		

6.5.2.5 Hold

6.5.2.5.1 Communication Hold with support for UPDATE

SS_XXSSCH01	NGN reference to: [21]				
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE				
Selection criteria:	Session hold. UPDATE method is a	used			
Test purpose:	Individual media streams are affected. The media stream was previously set to sendrecv.				
	Ensure that the IUT requesting the hold session stops sending media and sends an UPDATE to hold the session. Hold is done containing the SDP with the attribute "a=" sendonly. The IUT after requesting the hold session <i>receives</i> 200 OK final response containing the SDP with the attribute "a=" recvonly.				
Precondition:	 A session was established be 	etween	user A and user B according to the "basic Call"		
	procedures.				
	The media stream was previous	ously se	t to "sendrecv".		
	Individual media streams.				
SIP Parameter values:	Dial string parameters options=PIX	IT			
	TYPE_SDP= PIXIT;				
	PIXIT for supported header:				
	Case a) No 100 rel; Case b) Supported: 100 rel;				
	Case c) Supported: 100 rel, Case c) Supported: 100 rel ar	nd proce	andition		
Comments:	Case c) Supported. 100 fer al	iu precc	matton.		
SIP UA A	SUT		SIP UA B		
INVITE (sendrecv)	→	→	INVITE		
180 Ringing	+	É	180 Ringing		
200 OK INVITE	(-	200 OK INVITE		
UPDATE(sendonly)	→	→	UPDATE(sendonly)		
200 OK UPDATE (recvonly	y) ←	←	200 OK UPDATE (recvonly)		
BYE	→	→	BYE		
200 OK BYE	~	÷	200 OK BYE		

SS_XXSSCH02	NGN reference to: [21]				
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE				
Selection criteria:	Session hold. UPDATE method is used.				
Test purpose:	Individual media streams are affected. The media stream was previously set to recvonly.				
			ssion stops sending media and sends an		
			containing the SDP with the attribute "a="		
	containing the SDP with the attribut		session receives 200 OK final response		
Precondition:					
Frecondition.	Call" procedures.	etween	user A and user B according to the "basic		
	The media stream was prev	iouely e	et to "recyonly"		
	 Individual media streams. 	lously s	et to Tecvority.		
SIP Parameter values:	Dial string parameters options=PIX	IT			
on rarameter values.	TYPE_SDP= PIXIT;				
	PIXIT for supported header:				
	Case a) No 100 rel;				
	Case b) Supported: 100 rel;				
	Case c) Supported: 100 rel ar	nd preco	ondition.		
Comments:			OID IIA D		
SIP UA A	SUT	→	SIP UA B		
INVITE (sendrecv) 180 Ringing	→	→	180 Ringing		
200 OK INVITE	-	-	200 OK INVITE		
200 OK INVITE	•	•	200 OK INVITE		
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	+	É	200 OK BYE		

SS_XXSSCH03	NGN reference to: [21]				
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE				
Selection criteria:	Session hold. UPDATE method is used.				
Test purpose:	Individual media streams are affected. The media stream was previously set to sendonly.				
	_ ,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,				
			ne the session with user B the UE-A starts		
			resume the session with the attribute "a="		
			sting the hold session <i>receives</i> 200 OK final sendrecv in the SDP. The a=sendrecv attribute		
	is the default value therefore the				
Precondition:			n user A and user B according to the "basic		
r recondition.	Call" procedures.	Detweet	i user A and user B according to the basic		
	The media stream was pre	wiouely e	et to "sendonly"		
	 Individual media streams. 	sviously s	set to Seridonly .		
SIP Parameter values:	Dial string parameters options=PI	XIT			
on raidineter values.	TYPE_SDP= PIXIT;				
	PIXIT for supported header:				
	Case a) No 100 rel;				
	Case b) Supported: 100 rel;				
	Case c) Supported: 100 rel		ondition.		
Comments:					
SIP UA A	SUT		SIP UA B		
INVITE (sendrecv)	→	→	INVITE		
180 Ringing	((180 Ringing		
200 OK INVITE	←	←	200 OK INVITE		
UPDATE(sendonly)	→	→	UPDATE(sendonly)		
200 OK UPDATE (recvonly)	←	←	200 OK UPDATE (recvonly)		
LIDDATE (dura)	`		LIDDATE (see drees)		
UPDATE(sendrecv)	→ UPDATE(sendrecv) recv) ← 200 OK UPDATE (sendrecv)				
200 OK UPDATE (sendrecy	ecv) ← 200 OK UPDATE (sendrecv)				
BYE	→	→	BYE		
200 OK BYE	←	←	200 OK BYE		

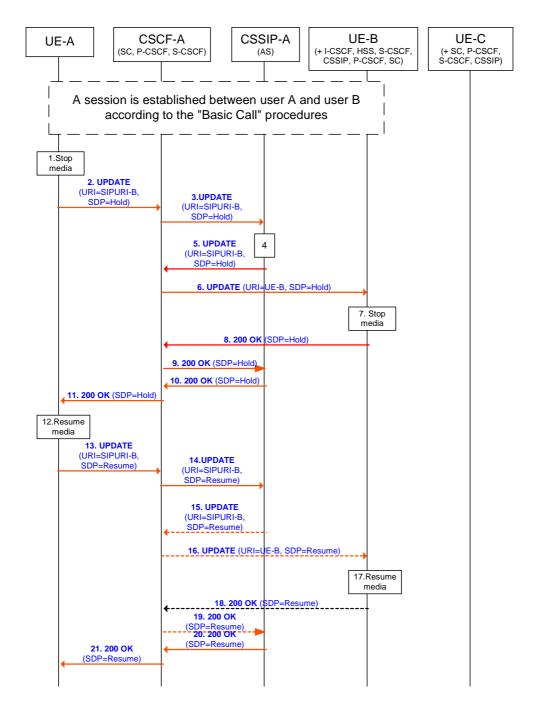
SS_XXSSCH04	NGN refere	nce to: [21]		
TSS reference:	ServedUser/With	outAnnounc/Withl	JPDATE	
Selection criteria:				dividual media streams are affected.
Test purpose:	Individual media streams are affected. The media stream was previously set to inactive.			
	Ensure that the IUT is requesting to resume the session with user B the UE-A starts			
	sending media and sends an UPDATE to resume the session with the attribute "a=" recvonly in the SDP.			
	The IUT after requesting the hold session <i>receives</i> 200 OK final response and optionally the attribute "a=" sendonly in the SDP.			
		•		
Precondition:			oetween	user A and user B according to the "basic Call"
	procedure			
		stream was prev	iously s	et to "inactive".
		media streams.		
SIP Parameter values:		eters options=PIX	IT	
	TYPE_SDP= PIX			
	PIXIT for support			
	Case a) No	,		
		ported: 100 rel;		10.0
	Case c) Sur	ported: 100 rel ar	nd preco	ondition.
Comments:	A			
SIP UA A		SUT		SIP UA B
INVITE (sendrecv)			→	INVITE
180 Ringing			(180 Ringing
200 OK INVITE	•	-	←	200 OK INVITE
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				
BYE		>	→	BYE
200 OK BYE		-	-	200 OK BYE

SS_XXSSCH05	NGN reference to: [21]		
TSS reference:	ServedUser/WithoutAnnounc/Withl	UPDATE	
Selection criteria:	Session hold. UPDATE method is a	used.	
Test purpose:	Individual media streams are affected. The media stream was previously set to sendrecv.		
		e hold session stops sending media and sends an	
	UPDATE to hold the session. Hold is done containing the SDP with the attribute "a="		
		g the hold session receives 200 OK final response	
	containing the SDP with the attribut		
Precondition:		between user A and user B according to the "basic	
	Call" procedures.		
	The media stream was prev		
	Media streams in the SDP.		
SIP Parameter values:	Dial string parameters options=PIX	KIT	
	TYPE_SDP= PIXIT;		
	PIXIT for supported header:		
	Case a) No 100 rel;		
	Case b) Supported: 100 rel; Case c) Supported: 100 rel ar	and precondition	
Comments:	Case c) Supported. 100 fer al	na precondition.	
SIP UA A	SUT	SIP UA B	
INVITE (sendrecv)	→	→ INVITE	
180 Ringing	É	← 180 Ringing	
200 OK INVITE	<u> </u>	€ 200 OK INVITE	
200 OK INVITE	•	200 OK INVITE	
UPDATE(sendonly)	→	UPDATE(sendonly)	
200 OK UPDATE (recvonly	←	← 200 OK UPDATE (recvonly)	
BYE	→	→ BYE	
200 OK BYE	←	€ 200 OK BYE	

SS_XXSSCH06	NGN reference to: [21]		
TSS reference:	ServedUser/WithoutAnnounc/WithI	JPDATI	Ε
Selection criteria:	Session hold. UPDATE method is used.		
Test purpose:	Individual media streams are affected. The media stream was previously set to recvonly.		
	Ensure that the IUT requesting the hold session stops sending media and sends an UPDATE to hold the session. Hold is done containing the SDP with the attribute "a=" inactive. The IUT after requesting the hold session <i>receives</i> 200 OK final response containing the SDP with the attribute "a=" inactive.		
Precondition:		oetween	user A and user B according to the "basic
	Call" procedures.		
	 The media stream was prev 	iously s	et to "sendonly".
	Media streams in the SDP.		
SIP Parameter values:	Dial string parameters options=PIX	.IT	
	TYPE_SDP= PIXIT;		
	PIXIT for supported header: Case a) No 100 rel;		
	Case b) Supported: 100 rel;		
	Case c) Supported: 100 rel ai	nd preco	ondition.
Comments:	,,		
SIP UA A	SUT		SIP UA B
INVITE (sendrecv)	→	→	INVITE
180 Ringing	←	←	180 Ringing
200 OK INVITE	←	+	200 OK INVITE
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	NGN reference to: [21]		
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE		
Selection criteria:	Session hold. UPDATE method is used. All the media streams are affected.		
Test purpose:	Individual media streams are affected. The media stream was previously set to sendonly.		
	Ensure that the IUT is requesting to resume the session with user B the UE-A starts sending media and sends an UPDATE to resume the session with the attribute "a=" sendrecv in the SDP The IUT after requesting the hold session <i>receives</i> 200 OK final response and optionally the attribute "a=" sendrecv in the SDP. The a=sendrecv attribute is the default value therefore the attribute can be omitted.		
Precondition:	 A session was established b 	etween	user A and user B according to the "basic
	Call" procedures.		
	 The media stream was previ 	iously s	et to "sendonly".
	 Media streams in the SDP. 		
SIP Parameter values:	Dial string parameters options=PIXI TYPE_SDP= PIXIT;	IT	
	PIXIT for supported header:		
	Case a) No 100 rel;		
	Case b) Supported: 100 rel;		
	Case c) Supported: 100 rel an	nd preco	ondition.
Comments:			
SIP UA A	SUT		SIP UA B
INVITE (sendrecv)	→	→	INVITE
180 Ringing	←	←	180 Ringing
200 OK INVITE	←	←	200 OK INVITE
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 200 OK BYE	→ ←	→	BYE 200 OK BYE

SS_XXSSCH08	NGN reference to: [21]	
TSS reference:	ServedUser/WithoutAnnounc/Withl	hUPDATE
Selection criteria:	Session hold. UPDATE method is a	s used. All the media streams are affected.
Test purpose:	All the media streams are affected.	ed. The media stream was previously set to sendonly.
	Ensure that the IUT is requesting to	to resume the session with user B the UE-A starts
		/ITE to resume the session with the attribute "a=" ter requesting the hold session receives 200 OK final
		oute "a=" sendrecv in the SDP. The a=sendrecv attribute
	is the default value therefore the at	
Precondition:		d between user A and user B according to the "basic
	Call" procedures.	
	 The media stream was prev 	
	Media streams in the SDP.	
SIP Parameter values:	Dial string parameters options=PIX	IXIT
	TYPE_SDP= PIXIT; PIXIT for supported header:	
	Case a) No 100 rel;	
	Case b) Supported: 100 rel;	
	Case c) Supported: 100 rel ar	and precondition.
Comments:		
SIP UA A	SUT	SIP UA B
INVITE (sendrecv)	→	→ INVITE ← 180 Ringing
180 Ringing 200 OK INVITE	~	← 180 Ringing← 200 OK INVITE
200 OK IIVVIIE	•	200 OK IIVITE
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



6.5.2.5.2 Communication Hold without support for UPDATE

SS_XXSSCH 09	NGN reference to: [21]		
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE		
Selection criteria:	UPDAT method is not supported, the reINVITE is used.		
Test purpose:	The media stream was previously set to sendrecv.		
	Ensure that in the case that UPDATE is not supported in one of the endpoints, and therefore does not have UPDATE in the allow header in the initial INVITE, Communication Hold should be done using a Re-INVITE. The UPDATE is not contained in Allow header in the 200 OK INVITE		
Precondition:	 A session was established between user A and user B according to the "basic Call" procedures. The media stream was previously set to "sendrecv". Individual media streams. 		

SIP Parameter values:	Dial string parameters options	=PIXIT		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:	PIXIT for supported header:		
	Case a) No 100 rel;			
	Case b) Supported: 100	rel;		
	Case c) Supported: 100		ondition.	
Comments:				
SIP UA A	SUT	Γ	SIP UA B	
INVITE (sendrecv)	→	→	INVITE	
180 Ringing	←	←	180 Ringing	
200 OK INVITE	←	←	200 OK ĬNVITE	
INVITE(sendonly)	→	→	INVITE(sendonly)	
200 OK INVITE (recvonly)	←	←	200 OK INVITE(recvonly)	
BYE	→	→	BYE	
200 OK BYE	←	←	200 OK BYE	

SS_XXSSCH 10	NGN reference to: [21]		
TSS reference:	ServedUser/WithoutAnnounc/Without	outUPDATE	
Selection criteria:	Session hold. INVITE method is used. Individual media streams are affected.		
Test purpose:	The media stream was previously set to sendrecv.		
	Ensure that the IUT requesting the hold session stops sending media and sends an INVITE to hold the session. Hold is done containing the SDP with the attribute "a=" sendonly. The IUT after requesting the hold session <i>receives</i> 200 OK final response containing the SDP with the attribute "a=" recvonly.		
Precondition:	 A session was established between user A and user B according to the "basic Call" procedures. The media stream was previously set to "sendrecv". Individual media streams. 		
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel al		
Comments: SIP UA A	SUT	SIP UA B	
INVITE (sendrecv)	→	→ INVITE	
180 Ringing 200 OK INVITE	*	← 180 Ringing← 200 OK INVITE	
INVITE(sendonly) 200 OK INVITE (recvonly)	→ ←	→ INVITE(sendonly)← 200 OK INVITE(recvonly)	
BYE 200 OK BYE	→	→ BYE ← 200 OK BYE	

SS_XXSSCH 11	NGN reference to: [21]		
TSS reference:	ServedUser/WithoutAnnounc/Without	tUPD	ATE
Selection criteria:	Session hold. INVITE method is used. Individual media streams are affected.		
Test purpose:	The media stream was previously set to sendrecv.		
	Ensure that the IUT requesting the hold session stops sending media and sends an INVITE to hold the session. Hold is done containing the SDP with the attribute "a=" sendonly. The IUT after requesting the hold session <i>receives</i> 200 OK final response containing the SDP with the attribute "a=" recvonly.		
Precondition:	 A session was established be 	tween	user A and user B according to the "basic
	Call" procedures.		
	The media stream was previous	usly s	et to "sendrecv".
	 Individual media streams. 		
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	preco	ondition.
Comments:	0.17		017 114 7
SIP UA A	SUT		SIP UA B
INVITE (sendrecv)	→ ←	→	INVITE
180 Ringing 200 OK INVITE	~	-	180 Ringing 200 OK INVITE
200 OK INVITE	•	_	200 OK INVITE
INVITE(sendonly)	→	→	INVITE(sendonly)
200 OK INVITE (recvonly)	-	-	200 OK INVITE(recvonly)
			, , , , , , , , , , , , , , , , , , , ,
BYE	→	→	BYE
200 OK BYE	←	←	200 OK BYE

SS_XXSSCH 12	NGN reference to: [21]			
TSS reference:	ServedUser/WithoutAnnounc/Withou	tUPD/	ATE	
Selection criteria:	Session hold. INVITE method is used. Individual media streams are affected			
Test purpose:	The media stream was previously set to recvonly.			
	INVITE to hold the session when . H	Ensure that the IUT requesting the hold session stops sending media and sends an INVITE to hold the session when . Hold is done containing the SDP with the attribute "a=" inactive. The IUT after requesting the hold session <i>receives</i> 200 OK final response		
	containing the SDP with the attribute	"a=" i	nactive.	
Precondition:	 A session was established be Call" procedures. 	tween	user A and user B according to the "basic	
	The media stream was previoIndividual media streams.	usly s	et to "recvonly".	
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;			
Comments:	Case c) Supported: 100 rel and precondition.			
SIP UA A	SUT SIP UA B			
INVITE (sendrecv)	→	→	INVITE	
180 Ringing	←	←	180 Ringing	
200 OK INVITE	←	←	200 OK INVITE	
INVITE (sendonly)	←	←	INVITE(sendonly)	
200 OK INVITE (recvonly)	→ 200 OK INVITE(recvonly)			
INVITE (inactive)	→	→	INVITE(inactive)	
200 OK INVITE (inactive)	← 200 OK INVITE(inactive)			
BYE	→	→	BYE	
200 OK BYE	+	+	200 OK BYE	

nedia stream was previously s re that the IUT is requesting to ng media and sends an INVIT	sed. Indi set to se o resum	vidual media streams are affected. endonly.
nedia stream was previously s re that the IUT is requesting to ng media and sends an INVIT	set to se	endonly.
re that the IUT is requesting to ng media and sends an INVIT	o resum	·
ng media and sends an INVIT		a the session with user R the LIE-A starts
Ensure that the IUT is requesting to resume the session with user B the UE-A starts sending media and sends an INVITE to resume the session with the attribute "a=" sendrecv in the SDP. The IUT after requesting the hold session <i>receives</i> 200 OK final response and optionally the attribute "a=" sendrecv in the SDP. The a=sendrecv attribute is the default value therefore the attribute can be omitted		
	oetween	user A and user B according to the "basic
	iously s	et to "sendonly".
	31	
,	nd preco	ondition.
, , , , , , , , , , , , , , , , , , , ,		
SUT		SIP
→	→	INVITE
		180 Ringing
←	+	200 OK INVITE
→	→	INVITE(sendonly)
←	←	200 OK INVITE(recvonly)
•		INDUITE (- an dec su)
-		INVITE (sendrecv) 200 OK INVITE (sendrecv)
•	•	200 OK HVVIIE (Sendiecv)
→	→	BYE
←	←	200 OK BYE
	default value therefore the at A session was established I Call" procedures. The media stream was preventional parameters options=PIXET; for supported header: ase a) No 100 rel; ase b) Supported: 100 rel; ase c) Supported: 100 rel at SUT SUT SUT	default value therefore the attribute of A session was established between Call" procedures. The media stream was previously string parameters options=PIXIT for supported header: ase a) No 100 rel; ase b) Supported: 100 rel; ase c) Supported: 100 rel and preconstructions of the construction of the constr

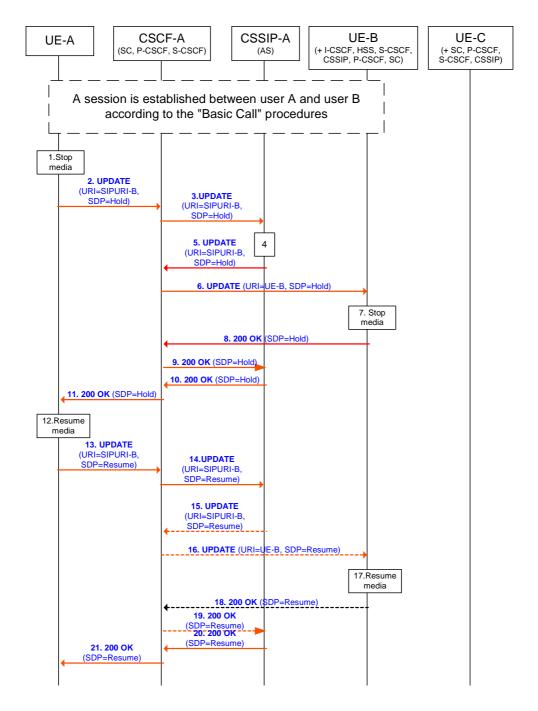
SS_XXSSCH 14	NGN reference to: [21]			
TSS reference:	ServedUser/WithoutAnnounc/With	noutUPD/	ATE	
Selection criteria:	Session hold. INVITE method is used. Individual media streams are affected.			
Test purpose:	The media stream was previously set to inactive.			
	Ensure that the IUT is requesting to resume the session with user B the UE-A starts sending media and sends an INVITE to resume the session with the attribute "a=" recvonly in the SDP. The IUT after requesting the hold session <i>receives</i> 200 OK final response and optionally the attribute "a=" sendonly in the SDP.			
Precondition:	A session was established between user A and user B according to the "basic Call" procedures. The media stream was previously set to "inactive".			
	 Individual media streams. 	,		
SIP Parameter values:	Dial string parameters options=PL TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel;		ondition.	
Comments:	, 11	-		
SIP UA A	SUT		SIP UA B	
INVITE (sendrecv)	→	→	INVITE	
180 Ringing	(←	180 Ringing	
200 OK INVITE	←	+	200 OK INVITE	
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)	
INVITE (recvonly)	→	→	INVITE (recvonly)	
200 OK INVITE (sendonly)	←	+	200 OK INVITE (sendonly)	
BYE	→	→	BYE	
200 OK BYE	*	←	200 OK BYE	

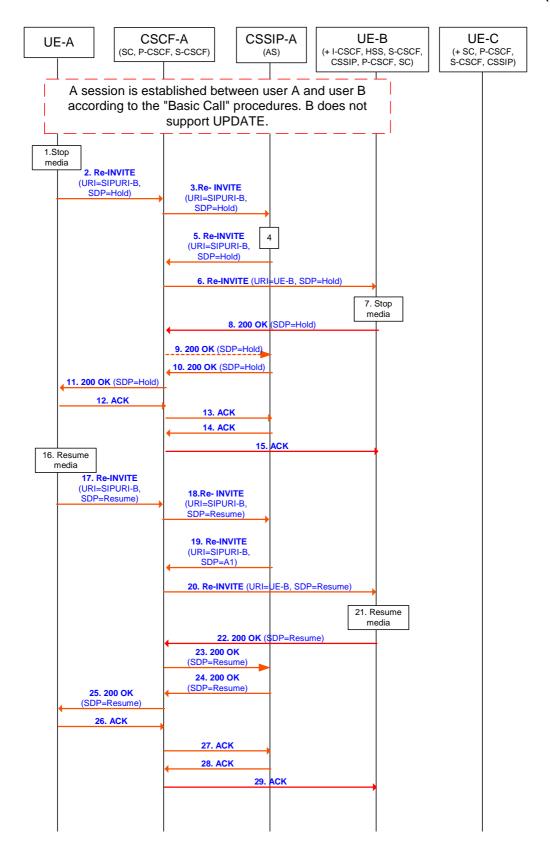
SS_XXSSCH 15	NGN reference to: [21]			
TSS reference:	ServedUser/WithoutAnnounc/Without	outUPDATE		
Selection criteria:	Session hold. INVITE method is used. All the media streams are affected.			
Test purpose:	The media stream was previously set to sendrecv.			
	Ensure that the IUT requesting the hold session stops sending media and sends an INVITE to hold the session. Hold is done containing the SDP with the attribute "a=" sendonly. The IUT after requesting the hold session <i>receives</i> 200 OK final response containing the SDP with the attribute "a=" recvonly.			
Precondition:	 A session was established between user A and user B according to the "basic Call" procedures. The media stream was previously set to "sendrecv". Media streams in the SDP. 			
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 UA A	SUT	SIP UA B		
INVITE (sendrecv)	→	→ INVITE		
180 Ringing	´	← 180 Ringing		
200 OK INVITE	← 200 OK INVITE			
INVITE(sendonly) 200 OK INVITE (recvonly)	→ ←	→ INVITE(sendonly)← 200 OK INVITE(recvonly)		
BYE 200 OK BYE	→	→ BYE← 200 OK BYE		

SS_XXSSCH 16	NGN refe	rence to: [21]				
TSS reference:	ServedUser/W	ServedUser/WithoutAnnounc/WithoutUPDATE				
Selection criteria:	Session hold.	Session hold. INVITE method is used. All the media streams.				
Test purpose:	are affected. T	are affected. The media stream was previously set to sendonly.				
	Ensure that the IUT is requesting to resume the session with user B the UE-					
			an INVITE to resume the session with the			
			P. The IUT after requesting the hold session			
			nd optionally the attribute "a=" sendrecv in the			
		endrecv attribute is	the default value therefore the attribute can be			
	omitted.					
Precondition:			between user A and user B according to the			
		Call" procedures.				
			eviously set to "recvonly".			
		streams in the SDP				
SIP Parameter values:		ameters options=PI	XIT			
	TYPE_SDP= F					
	PIXIT for supp					
	Case a)					
		Supported: 100 rel;				
0	Case c)	Supported: 100 rel	and precondition.			
Comments: SIP UA A		SUT	SIP UA B			
INVITE (sendrecv)	→	→	INVITE			
180 Ringing	É	É	180 Ringing			
200 OK INVITE	÷	÷	200 OK INVITE			
200 011 111112	-	•	200 01(1144112			
INVITE (sendonly)	←	←	INVITE(sendonly)			
200 OK INVITE (recvonly)	→	→	200 OK INVITE(recvonly)			
` **			(
INVITE(inactive)	→ INVITE(inactive)					
200 OK INVITE (inactive)	← 200 OK INVITE(inactive)					
·						
BYE	→ → BYE					
200 OK BYE	←	←	200 OK BYE			

SS_XXSSCH 17	NGN reference to: [21]			
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE			
Selection criteria:	Session hold. INVITE method is used. All the media streams are affected.			
Test purpose:	The media stream was previously	The media stream was previously set to sendonly.		
	Ensure that the IUT is requesting to resume the session with user B the UE-A starts sending media and sends an INVITE to resume the session with the attribute "a=" sendrecv in the SDP. The IUT after requesting the hold session <i>receives</i> 200 OK final response and optionally the attribute "a=" sendrecv in the SDP. The a=sendrecv attribute is the default value therefore the attribute can be omitted.			
Precondition:		betweer	n user A and user B according to the "basic	
	Call" procedures.The media stream was prev	iouch c	sot to "sandanly"	
	 Media streams in the SDP. 	riousiy s	sector sericonly.	
SIP Parameter values:	Dial string parameters options=PIX	/IT		
on ranameter values.	TYPE_SDP= PIXIT;	KI I		
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel.			
	Case c) Supported: 100 rel a	nd prec	ondition.	
Comments:				
SIP UA A	SUT	_	SIP UA B	
INVITE (sendrecv)	→	→	INVITE	
180 Ringing	((180 Ringing	
200 OK INVITE	←	+	200 OK INVITE	
INVITE (sendonly)	→	→	INVITE(sendonly)	
200 OK INVITE (recvonly)	←	←	200 OK INVITE(recvonly)	
` ,				
INVITE (sendrecv)	→ INVITE (sendrecv)			
200 OK INVITE (sendrecv)	← 200 OK INVITE (sendrecv)			
BYE	→	→	BYE	
200 OK BYE	←	←	200 OK BYE	

SS_XXSSCH 18	NGN reference to: [21]			
TSS reference:	ServedUser/WithoutAnnounc/With			
Selection criteria:	Session hold. INVITE method is us	sed. All	the media streams are affected.	
Test purpose:	The media stream was previously set to inactive.			
	Ensure that the IUT is requesting to resume the session with user B the UE-A starts			
	sending media and sends an INVITE to resume the session with the attribute "a=" recvonly in the SDP. The IUT after requesting the hold session <i>receives</i> 200 OK final response and optionally the attribute "a=" sendonly in the SDP.			
Precondition:				
Precondition.	Call" procedures.		n user A and user B according to the "basic	
	 The media stream was prev 	iously s	set to "inactive".	
	 Media streams in the SDP. 			
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT;	ΊΤ		
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;			
	Case c) Supported: 100 rel a	nd prec	ondition.	
Comments:				
SIP UA A	SUT	_	SIP	
INVITE (sendrecv))	→	INVITE	
180 Ringing	((180 Ringing	
200 OK INVITE	←	(200 OK INVITE	
INVITE(sendonly)	←	←	INVITE(sendonly)	
200 OK INVITE (recvonly)	→	→	200 OK INVITE(recvonly)	
INVITE(inactive))	→	INVITE(inactive)	
200 OK INVITE (inactive)	← 200 OK INVITE(inactive)			
INVITE (recvonly)	→ INVITE (recvonly)			
200 OK INVITE (sendonly)	← 200 OK INVITE (sendonly)			
ACK	→ ACK			
BYE	→ BYE			
200 OK BYE	←	←	200 OK BYE	





6.5.2.5.3 Communication with announcements

6.5.2.5.3.1 Communication Hold with support for UPDATE

SS_XXSSCH 19	NGN reference to: [21]			
TSS reference:	ServedUser/WithAnnounc/WithUPDAT	ΓΕ		
Selection criteria:	The emote user is put on hold, an announcement starts to the held user. The UPDATE method is used. Individual media streams are affected.			
Test purpose:	Ensure that when the remote user is s remote UE. An UPDATE is sent to the		n HOLD, an announcement is started to the ote user B with SDP a=sendonly.	
Precondition:	 A session was established between user A and user B according to the "basic Call" procedures. Individual media streams. 			
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:	015			
SIP UA A	SUT	_	SIP UA B	
INVITE (sendrecv)	→	→	INVITE	
180 Ringing	←	←	180 Ringing	
200 OK INVITE	←	←	200 OK INVITE	
UPDATE(sendonly)	→ UPDATE(sendonly)			
200 OK UPDATE (recvonly)	`			
,,	Announcement to UE B			
BYE	→ → BYE			
200 OK BYE	←	←	200 OK BYE	

SS_XXSSCH 20	NGN reference to: [21]				
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. Individua				
Test purpose:	Ensure that no announcement is sta	arted to	any user if the media stream was previously		
		tly put	on hold by user B. An UPDATE is sent to User		
	A with a SDP a=inactive.				
Precondition:		etweer	n user A and user B according to the "basic		
	Call" procedures.				
	Individual media streams.				
SIP Parameter values:	Dial string parameters options=PIXI	I			
	TYPE_SDP= PIXIT; PIXIT for supported header:				
	Case a) No 100 rel;				
	Case b) Supported: 100 rel;				
	Case c) Supported: 100 rel an	d prec	ondition.		
Comments:					
SIP UA A	SUT		SIP UA B		
INVITE (sendrecv)	→	→	INVITE		
180 Ringing	←	←	180 Ringing		
200 OK INVITE	←	←	200 OK INVITE		
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 sream is stopped					
BYE	→	→	BYE		
200 OK BYE	←	←	200 OK BYE		

SS_XXSSCH 21	NGN reference to: [21]				
TSS reference:	ServedUser/WithAnnounc/WithUPD	ATE			
Selection criteria:	The announcement is stopped after retrieve. Individual media streams are affected.				
Test purpose:	Ensure that the announcement started to user B is stopped when the user B is retrieved				
	by user A. An UPDATE is sent with SDP a=sendrecv. The normal conversation shall				
		apply between user A and user B. The a=sendrecv attribute is the default value therefore			
	the attribute can be omitted.				
Precondition:		etween	user A and user B according to the "basic		
	Call" procedures.				
	 Individual media streams. 				
SIP Parameter values:	Dial string parameters options=PIXI	T			
	TYPE_SDP= PIXIT;				
	PIXIT for supported header:				
	Case a) No 100 rel;				
	Case b) Supported: 100 rel;	d	and distant		
Comments:	Case c) Supported: 100 rel an	a prece	ondition.		
SIP UA A	SUT		SIP UA B		
INVITE (sendrecv)	→	→	INVITE		
180 Ringing	←	-	180 Ringing		
200 OK INVITE	←	←	200 OK INVITE		
UPDATE(sendonly)	→	→	UPDATE(sendonly)		
200 OK UPDATE (recvonly)	←	←	200 OK UPDATE (recvonly)		
	Announcement to UE B				
UPDATE(sendrecv)	→ UPDATE(sendrecv)				
200 OK UPDATE (sendrecv)					
Conversation					
BYE	→	→	BYE		
200 OK BYE	←	←	200 OK BYE		

SS_XXSSCH 22	NGN re	ference to: [21]		
TSS reference:	ServedUser/	WithAnnounc/WithUPI	DATE	
Selection criteria:	Announcement is started to user B when user B retrieves the connection. Individual media streams are affected.			
Test purpose:	Ensure that when user B retrieves the connection and is still held by the user A (was previously set on hold by user A), an UPDATE with SDP a=recvonly is sent to user A, the announcement is started to user B.			
Precondition:	 A session was established between user A and user B according to the "basic Call" procedures. Individual media streams. 			
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 UA A		SUT		SIP UA B
INVITE (sendrecv) 180 Ringing		→	→	INVITE 180 Ringing
200 OK INVITE		(-	200 OK INVITE
UPDATE(sendonly) 200 OK UPDATE (recvonly Announcement to		→	→	UPDATE(sendonly) 200 OK UPDATE (recvonly)
UPDATE(inactive)	OE B	←	+	UPDATE(inactive)
200 OK UPDATE (inactive) Media sream is sto	ve) → 200 OK UPDATE (inactive)			,
UPDATE(recvonly)	← ← UPDATE(recvonly)			
200 OK UPDATE (sendonly Announcement to	nly) → 200 OK UPDATE (sendonly)			
BYE 200 OK BYE		→	→	BYE 200 OK BYE

SS_XXSSCH 23	NGN reference to: [21]				
TSS reference:	ServedUser/WithAnnounc/WithUF	ServedUser/WithAnnounc/WithUPDATE			
Selection criteria:	The emote user is put on hold, an announcement starts to the held user. The UPDATE				
	method is used. All the media stre	eams are	e affected.		
Test purpose:			n HOLD, an announcement is started to the		
	remote UE. An UPDATE is sent to	o the rem	note user with SDP a=sendonly.		
Precondition:	 A session was established 	l betweer	n user A and user B according to the "basic		
	Call" procedures.				
	 Media streams in the SDP 				
SIP Parameter values:	Dial string parameters options=PI	XIT			
	TYPE_SDP= PIXIT;				
	PIXIT for supported header:				
	Case a) No 100 rel;				
	Case b) Supported: 100 rel;				
	Case c) Supported: 100 rel	and prec	ondition.		
Comments:					
SIP UA A	, SUT	_	SIP UA B		
INVITE (sendrecv)	→	→	INVITE		
180 Ringing	-	(180 Ringing		
200 OK INVITE Conversation	~	~	200 OK INVITE		
	_	_	LIDDATE/condonly)		
UPDATE(sendonly) 200 OK UPDATE (recvonly)	→ UPDATE(sendonly) ← 200 OK UPDATE (recvonly)				
Announcement to U					
Announcement to	JE B				
BYE	→ → BYF				
200 OK BYE	-	←	200 OK BYE		
	-	=			
•					

SS_XXSSCH 24	NGN reference to: [21]			
TSS reference:	ServedUser/WithAnnounc/WithUP	ServedUser/WithAnnounc/WithUPDATE		
Selection criteria:	The announcement is stopped after the held user puts the media stream on hold. The			
	UPDATE method is used. All the media streams are affected.			
Test purpose:		tarted to any user if the media stream was prevviously		
	put on hold by user A is subsequently put on hold by user B. An UPDATE is sent to User			
	A with a SDP a=inactive.			
Precondition:		between user A and user B according to the "basic		
	Call" procedures.			
	Media streams in the SDP.			
SIP Parameter values:	Dial string parameters options=PIX	(IT		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;			
Commonto	Case c) Supported: 100 rel a	na precondition.		
Comments: SIP UA A	SUT	SIP UA B		
INVITE (sendrecv)	• •	→ INVITE		
180 Ringing	,	← 180 Ringing		
200 OK INVITE	È	€ 200 OK INVITE		
200 010 1140 112	•	200 OK IIVITE		
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 sream is stopped			
BYE	→	→ BYE		
200 OK BYE	÷	€ 200 OK BYE		

SS_XXSSCH 25	NGN reference to: [21]			
TSS reference:	ServedUser/WithAnnounc/WithUPDATE			
Selection criteria:	Announcement is stopped after retrieve. All the media streams are affected.			
Test purpose:	Ensure that the announcement started to user B is stopped when the user B is retrieved by user A. An UPDATE is sent with SDP a=sendrecv. The normal conversation shall apply between user A and user B. The a=sendrecv attribute is the default value therefore the attribute can be omitted.			
Precondition:	 A session was established between user A and user B according to the "basic Call" procedures. Media streams in the SDP. 			
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 UA A	SUT		SIP UA B	
INVITE (sendrecv)	→	→	INVITE	
180 Ringing	<	(
200 OK INVITE	←	(200 OK INVITE	
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	

SS_XXSSCH 26	NGN reference to:	[21]		
TSS reference:	ServedUser/WithAnnoun	c/WithUPDATE		
Selection criteria:	Announcement is started to user B when user B retrieves the connection. All the media			
	streams are affected.			
Test purpose:			nnection and is still held by the user A (was	
	previously set on hold by user A), an UPDATE with SDP a=recvonly is sent to user A, the			
	announcement is started			
Precondition:		ablished betwee	n user A and user B according to the "basic Call"	
	procedures.			
	Media streams in			
SIP Parameter values:	Dial string parameters op	tions=PIXII		
	TYPE_SDP= PIXIT;	or.		
	PIXIT for supported head Case a) No 100 rel;			
	Case b) Supported:			
		100 rel and pred	condition	
Comments:	Cass of Capported.	100 for and proc	orialion.	
SIP UA A		SUT	SIP UA B	
INVITE (sendrecv)	→	→	INVITE	
180 Ringing	←	←	180 Ringing	
200 OK INVITE	←	←	200 OK INVITE	
UPDATE(sendonly)	→	→	UPDATE(sendonly)	
200 OK UPDATE (recvonly)	←	+	200 OK UPDATE (recvonly)	
LIBBATE (; , ;)	•	_	LIDDATE (; , ;)	
UPDATE(inactive)	-	+	UPDATE(inactive)	
200 OK UPDATE (inactive)	→ 200 OK UPDATE (inactive)			
UPDATE(recvonly)	← ← UPDATE(recvonly)			
200 OK UPDATE (sendonly)	÷	÷	200 OK UPDATE (sendonly)	
Announcement to UE B				
	_			
BYE	→ BYE			
200 OK BYE	←	+	200 OK BYE	

6.5.2.5.3.2 Communication Hold without support for UPDATE

SS_XXSSCH 27	NGN reference to: [21]		
TSS reference:	ServedUser/WithAnnounc/WithoutUPDATE		
Selection criteria:	The emote user is put on hold, an announcement starts to the held user. The INVITE method is used. Individual media streams are affected.		
Test purpose:	Ensure that when the remote user is set on HOLD, an announcement is started to the remote UE. An INVITE is sent to the remote user with SDP a=sendonly.		
Precondition:	 A session was established between user A and user B according to the "basic Call" procedures. Individual media streams. 		
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel ai		ndition.
Comments:			
SIP UA A	SUT	→	SIP UA B
INVITE (sendrecv)	-	-	INVITE
180 Ringing 200 OK INVITE	+	-	180 Ringing 200 OK INVITE
INVITE(sendonly) 200 OK INVITE (recvonly)	→ ← Announce	→ ← ment to U	INVITE (sendonly) 200 OK INVITE (recvonly) JE B
BYE 200 OK BYE	→	→	BYE 200 OK BYE

SS_XXSSCH 28	NGN reference to: [21]		
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. Individual media streams are affected.		
Test purpose:	Ensure that no announcement is started to any user if the media stream was previously put on hold by user A is subsequently put on hold by user B. An INVITE is sent to User A with a SDP a=inactive.		
Precondition:	 A session was established between user A and user B according to the "basic Call" procedures. Individual media streams. 		
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 precor	ndition.	
Comments:			
SIP UA A	SUT	SIP UA B	
INVITE (sendrecv)	*	INVITE	
180 Ringing 200 OK INVITE	+ + +	180 Ringing 200 OK INVITE	
INVITE(sendonly)	→ →	INVITE (sendonly)	
200 OK INVITE (recvonly)	← ← ← Announcement to UE B	200 OK INVITE(recvonly)	
INVITE (inactive)	+ +	INVITE (inactive)	
200 OK INVITE (inactive)	→ →	200 OK INVITE (inactive)	
Media sream is stopped			
BYE	→ →	BYE	
200 OK BYE	+ +	200 OK BYE	

SS_XXSSCH 29	NGN reference to: [21]		
TSS reference:	ServedUser/WithAnnounc/WithoutUPDATE		
Selection criteria:	Announcement is stopped after retrieve. The INVITE method is used. Individual media streams are affected.		
Test purpose:	Ensure that the announcement started to user B is stopped when the user B is retrieved by user A. An INVITE is sent with SDP a=sendrecv. The normal conversation shall apply between user A and user B. The a=sendrecv attribute is the default value therefore the attribute can be omitted.		
Precondition:	 A session was established between user A and user B according to the "basic Call" procedures. Individual media streams. 		
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 UA A	SUT	SIP UA B	
INVITE (sendrecv)	→ →		
180 Ringing 200 OK INVITE	+ + +	180 Ringing 200 OK INVITE	
INVITE(sendonly)	> >	INVITE(sendonly)	
200 OK INVITE (recvonly)	← ← 200 OK INVITE (recvonly) Announcement to UE B		
INVITE (sendrecv)	→ →	(
200 OK INVITE (sendrecv)	← ← ← Conversation	200 OK INVITE (sendrecv)	
BYE	→ →	BYE	
200 OK BYE	*	200 OK BYE	

ServedUser/WithAnnounc/Witho	outUPDATE	
Announcement is started to user B when user B retrieves the connection. The INVITE		
method is used. Individual media streams are affected.		
Ensure that when user B retrieves the connection and is still held by the user A (was previously set on hold by user A), an INVITE with SDP a=recvonly is sent to user A, the		
 A session was established between user A and user B according to the "basic 		
Dial string parameters options=F	PIXIT	
,		
Case c) Supported: 100 re	and precondition.	
CUT	CID IIA D	
	SIP UA B → INVITE	
	← 180 Ringing	
	€ 200 OK INVITE	
•	200 OK HVVIIE	
→	→ INVITE(sendonly)	
É	← 200 OK INVITE(recvonly)	
	(** * * *)	
←	← INVITE(inactive)	
→	→ 200 OK INVITE (inactive)	
←	INVITE (recvonly)	
	→ 200 OK INVITE (sendonly)	
Announceme	nt to UE B	
→	→ BYE	
´	€ 200 OK BYE	
	Announcement is started to use method is used. Individual media Ensure that when user B retriev previously set on hold by user A announcement is started to user • A session was established Call" procedures. • Individual media streams Dial string parameters options=1 TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rectors Case c) Supported: 100 rectors Supported: 100 rectors Case c) SUT Announcement	

SS_XXSSCH 31	NGN reference to: [21]		
TSS reference:	ServedUser/WithAnnounc/Without	UPDATE	
Selection criteria:	The emote user is put on hold, an announcement starts to the held user. The INVITE method is used. All the media streams are affected.		
Test purpose:	Ensure that when the remote user is set on HOLD, an announcement is started to the remote UE. An INVITE is sent to the remote user with SDP a=sendonly.		
Precondition:	 A session was established between user A and user B according to the "basic Call" procedures. Media streams in the SDP. 		
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel a		ndition.
Comments:			
SIP UA A	SUT		SIP UA B
INVITE (sendrecv)	→	→	INVITE
180 Ringing 200 OK INVITE	← ←	+	180 Ringing 200 OK INVITE
INVITE(sendonly) 200 OK INVITE (recvonly)	→ ← Announcement	→ ← to UE B	INVITE (sendonly) 200 OK INVITE (recvonly)
BYE 200 OK BYE	→ ←	→	BYE 200 OK BYE

SS_XXSSCH 32	NGN reference to: [21]		
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. All the media streams are affected.		
Test purpose:	Ensure that no announcement is started to any user if the media stream was previously		
	put on hold by user A is subsequently put on hold by user B. An INVITE is sent to User A		
	with a SDP a=inactive.		
Precondition:	 A session was established between user A and user B according to the "basic 		
	Call" procedures.		
	Media streams in the SDP.		
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; Case c) Supported: 100 rel and prec	ondition	
Comments:	eact of Cappented. For its and pro-	ondition.	
SIP UA A	SUT	SIP UA B	
INVITE (sendrecv)	→ →	INVITE	
180 Ringing	+ +	180 Ringing	
200 OK INVITE	+ +	200 OK INVITE	
	_		
INVITE(sendonly)	→ →	INVITE (sendonly)	
200 OK INVITE (recvonly)	← ← Announcement to UE I	200 OK INVITE(recvonly)	
INVITE (inactive)	Announcement to ∪E i	INVITE (inactive)	
200 OK INVITE (inactive)	5 5	200 OK INVITE (inactive)	
200 OK IIVITE (mactive)	Media sream is stoppe		
	oa.a o. oa lo otoppo		
BYE	→ →	BYE	
200 OK BYE	+ +	200 OK BYE	

SS_XXSSCH 33	NGN reference to: [21]			
TSS reference:	ServedUser/WithAnnounc/WithoutU	JPDATE		
Selection criteria:				
Test purpose:	Announcement is stopped after retrieve. The INVITE method is used. All the media			
	streams are affected.			
	Ensure that the announcement started to user B is stopped when the user B is retrieved			
	by user A. An INVITE is sent with SDP a=sendrecv. The normal conversation shall apply			
	between user A and user B. The a=sendrecv attribute is the default value therefore the			
	attribute can be omitted.			
Precondition:		etween	user A and user B according to the "basic	
	Call" procedures.			
	Media streams in the SDP.			
SIP Parameter values:	Dial string parameters options=PIX	IT		
	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel; Case c) Supported: 100 rel ar	nd nraco	ndition	
Comments:	edac e) edpported. 100 fer di	ia prece	ndition.	
SIP UA A	SUT		SIP UA B	
INVITE (sendrecv)	→	→	INVITE	
180 Ringing	←	←	180 Ringing	
200 OK INVITE	←	←	200 OK INVITE	
INVITE(sendonly)	→	→	INVITE(sendonly)	
200 OK INVITE (recvonly)	←	←	200 OK INVITE (recvonly)	
	Announcement t			
INVITE (sendrecv)	→	→	INVITE (sendrecv)	
200 OK INVITE (sendrecv)	←	←	200 OK INVITE (sendrecv)	
Conversation				
BYE	→	→	BYE	
200 OK BYE	←	←	200 OK BYE	

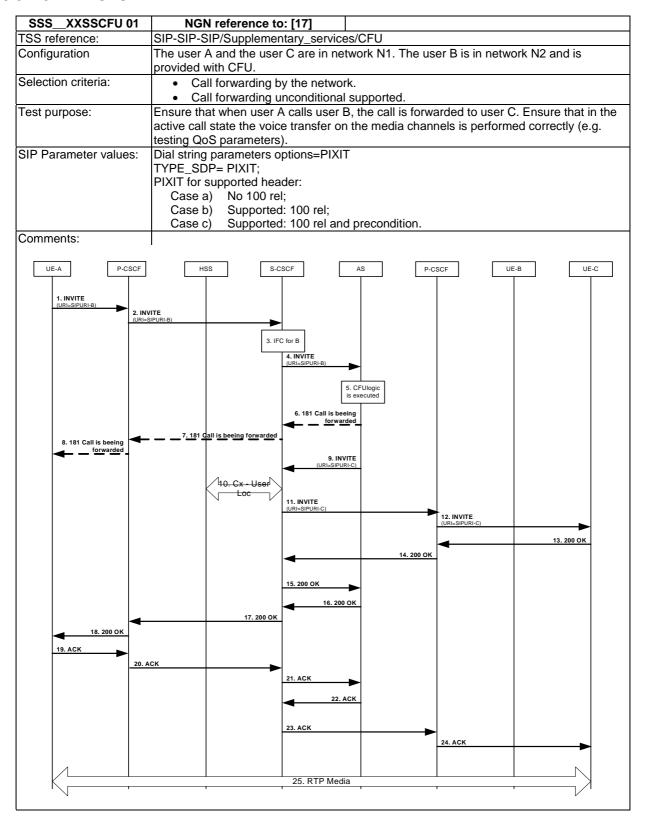
SS_XXSSCH 32	NGN reference to: [21]		
	ServedUser/WithAnnounc/Without		
	Announcement is started to user B when user B retrieves the connection. The INVIT method id used. All the media streams are affected.		
	Ensure that when user B retrieves the connection and is still held by the user A (was previously set on hold by user A), an INVITE with SDP a=recvonly is sent to user A, the announcement is started to user B.		
Precondition:	 A session was established between user A and user B according to the "basic Call" procedures. Media streams in the SDP. 		
	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel a		ndition.
Comments:			
SIP UA A	SUT	_	SIP UA B
INVITE (sendrecv)	→	→	INVITE
180 Ringing 200 OK INVITE	,	+	180 Ringing 200 OK INVITE
INVITE(sendonly)	→	→	INVITE(sendonly)
200 OK INVITE (recvonly)		←	200 OK INVITE(recvonly)
INVITE (inactive)	←	+	INVITE(inactive)
200 OK INVITE (inactive)	→	→	200 OK INVITE(inactive)
INVITE (recvonly)	←	+	INVITE (recvonly)
200 OK INVITE (sendonly		→	200 OK INVITE (sendonly)
(22)	Announceme	ent to UE	
BYE	→	→	BYE
200 OK BYE	←	-	200 OK BYE

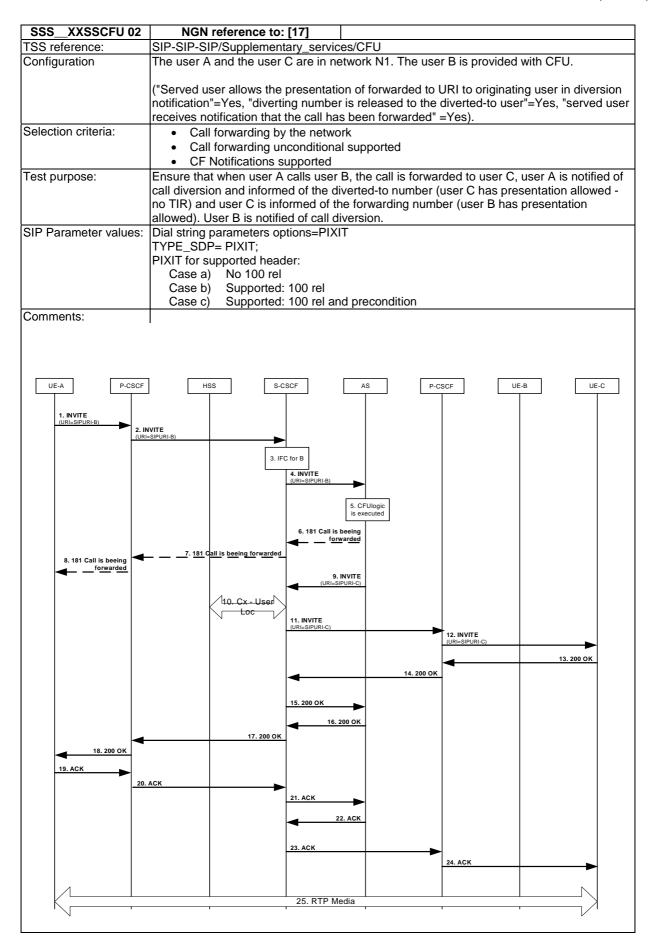
6.5.2.5.3.3 The early dialogue is established

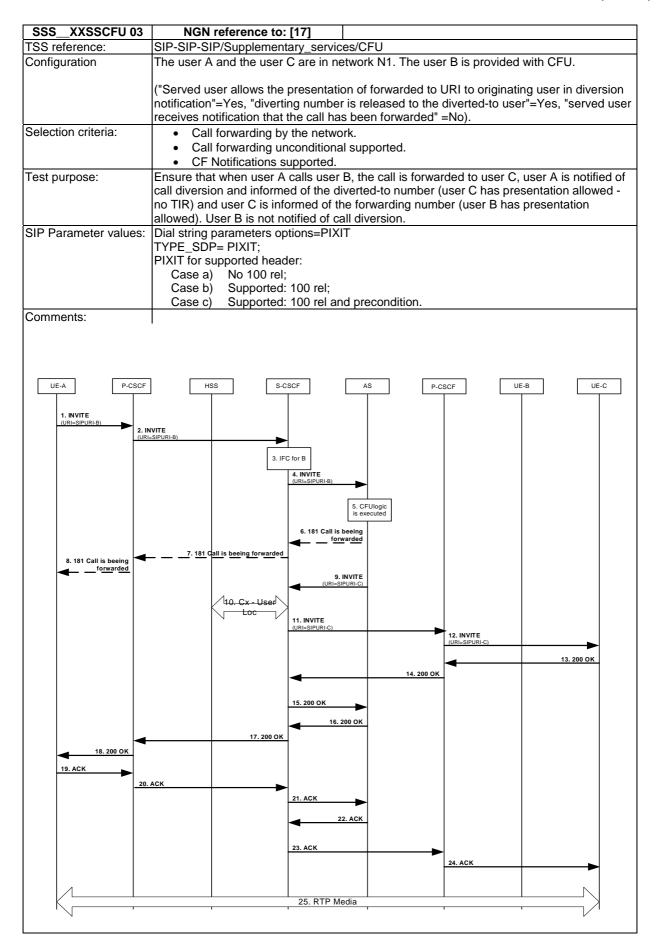
SS_XXSSCH 33	NGN reference to: [21]		
TSS reference:	ServedUser/RingingState		
Selection criteria:			
Test purpose:	Hold is done containing the SDP w	h <mark>old the</mark> : <i>i</i> ith the at	he early dialogue is established. session sends an UPDATE to hold the session. tribute "a=" sendonly. The IUT after requesting TE) message containing the SDP with the
Precondition:	Early dialogue was established be procedures.	tween use	er A and user B according to the "basic Call"
SIP Parameter values:	Dial string parameters options=PIXTYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel Case b) Supported: 100 rel Case c) Supported: 100 rel a		ndition
Comments: SIP UA A INVITE (sendrecv) 180 Ringing	SUT → ←	→	SIP UA B INVITE 180 Ringing
UPDATE(sendonly) 200 OK UPDATE (recvor	nly) →	→	UPDATE(sendonly) 200 OK UPDATE (recvonly)
BYE 200 OK BYE	→	→	BYE 200 OK BYE

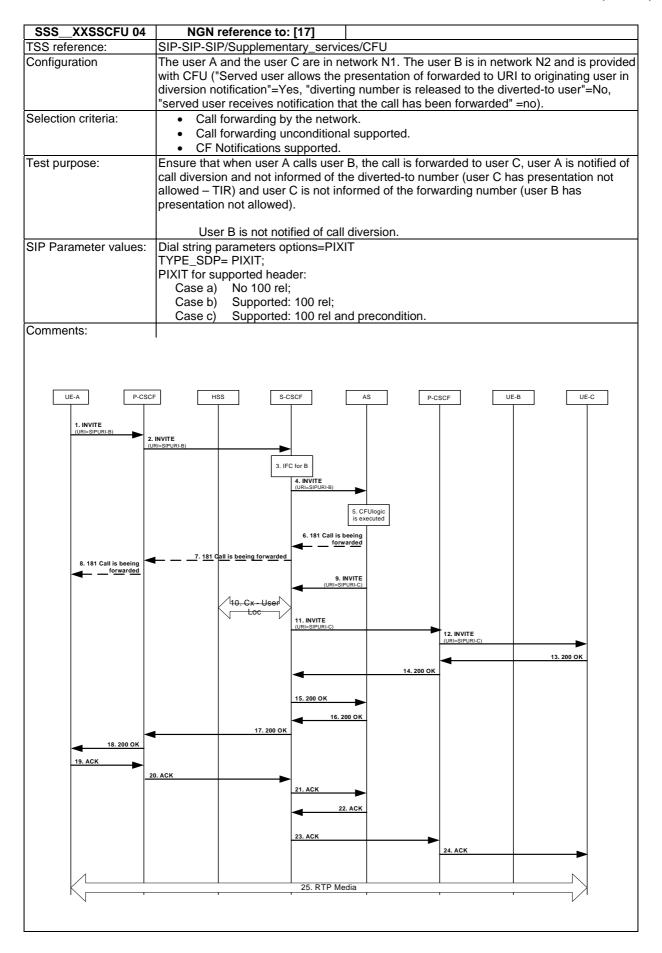
SS_XXSSCH 34	NGN reference to: [21]		
TSS reference:	ServedUser/RingingState		
Selection criteria:			
Test purpose:	The caller retrieves the media stre	am; the e	early dialogue is established.
	Ensure that the IUT is requesting to resume the session with user B the UE-A starts sending media and sends a UPDATE to resume the session with the supported codec in the SDP. The IUT after requesting the hold session receives a 200 OK (UPDATE) message containing the SDP with the supported codec. The a=sendrecv attribute is the default value therefore the attribute can be omitted.		
Precondition:	session was established between procedures	user A ar	nd user B according to the "basic Call"
SIP Parameter values:	Dial string parameters options=PIXTYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel a		ndition.
Comments: SIP UA A	SUIT		CID HA D
INVITE (sendrecv)	SUT	→	SIP UA B
180 Ringing	(-	180 Ringing
UPDATE(sendonly) 200 OK UPDATE (recvo	only) →	→	UPDATE(sendonly) 200 OK UPDATE (recvonly)
UPDATE(sendrecv) 200 OK UPDATE (send	recv) →	→	UPDATE(sendrecv) 200 OK UPDATE (sendrecv)
BYE 200 OK BYE	→ ←	→	BYE 200 OK BYE

6.5.2.6 CFU

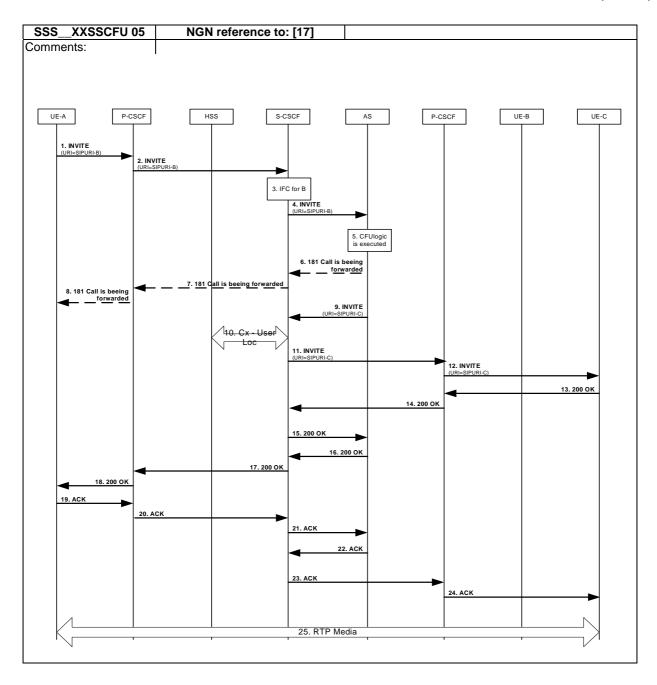


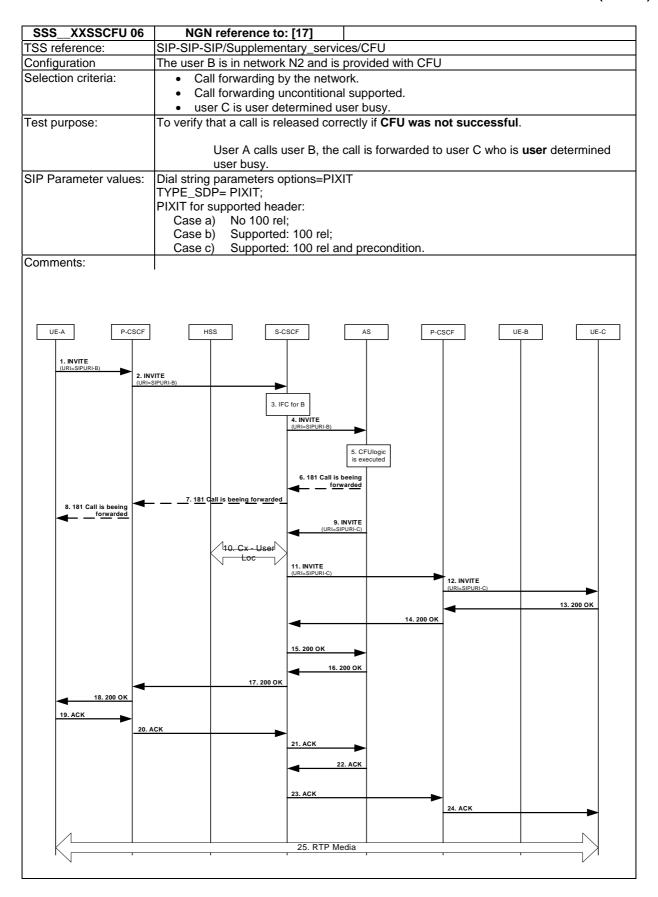


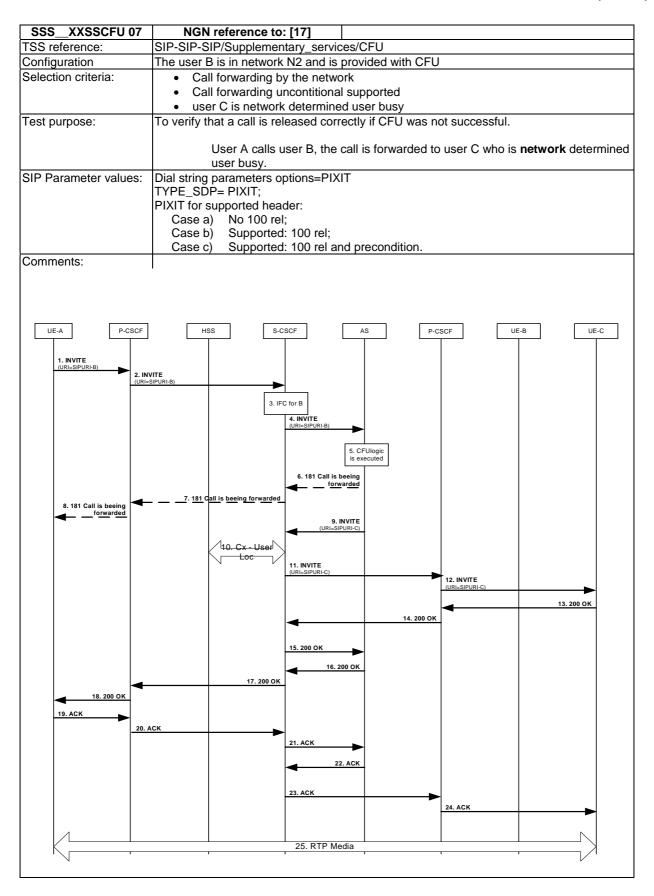




SSS_XXSSCFU 05	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFU
Configuration	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFU.
	The user A and the user C are in network N1 and user C is provided with TIP. The user B is in network N2 and is provided with CFU ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=No, "diverting number is released to the diverted-to user"=No, "served user receives notification that the call has been forwarded" =no).
Selection criteria:	 Call forwarding by the network. Call forwarding unconditional supported. CF Notifications supported.
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is not notified of call diversion and not informed of the diverted-to number (user C has TIP) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion.
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.







SSS XXSSCFU 08	NGN reference to:		
	TS 183 004 [17] clause 4.5.2.6.3		
TSS reference:	SIP-SIP-SIP/Supplementary_services	/CFU	
Configuration	The user B is in network N2 and is pro	ovided with CFU.	
Selection criteria:	 Call forwarding by the network 		
	 Call forwarding uncontitional s 	upported.	
	 user C is network determined 		
Test purpose:	The served user subscribes to the CF		ginating user is notified.
	Additionaly an announcement is playe	ed.	
	When Communication Diversion occuoriginating user is supported then a 1 towards the originating user.	81 (Call Is Being Forward	ed) response shall be sent
	Additional the AS initiates an annound		
	order to inform about the diversion. A		layed according to
OID D	procedures as are described in TS 18	3 028 [18].	
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.	
	181 Call is Being Forwarded: P-Asser		
Comments:			
SIP#1	SUT	SIP#2 (served user)	SIP#3
INVITE 1	→	,	
181 Call is Being Forwar	INVIT 180 Rin	E 2 →	→ INVITE ← 180 Ringing
180 Ringing	← 180 Ringing 200 OK (INV)	TE\ 4	← 200 OK (INVITE)
200 OK (INVITE)	€ 200 OK (INVITE)	1 L / L	200 01 (111112)
ACK	→ 200 OK (IIVITE)		→ ACK
	Communicat	on	
BYE	→		→ BYE
200 OK (BYE)	←		← 200 OK (BYE)
The test case can not l	be tested with an ATS		

SSS_XXSSCFU 09	NGN reference to:		
	TS 183 004 [17] clause 4.5.2.6.3		
TSS reference:	SIP-SIP-SIP/Supplementary_services		
Configuration	The user B is in network N2 and is pro	ovided with CFU	
Selection criteria:	 Call forwarding by the network 		
	 Call forwarding uncontitional services 	upported	
	user C is network determined to	user busy	
Test purpose:	The served user subscribes to the CF	· · · · · · · · · · · · · · · · · · ·	
	diversion to the diverting user using the	ne MESSAGE request trigge	red by a timer.
	Ensure that when the diverting user is		
	the information where the call is diver- timer value that can be provided by th		est that is be sent due to an
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		
	MESSAGE triggerd by a timer value p	provided the served user.	
Comments:			
SIP#1	SUT	SIP#2 (served user)	SIP#3
INVITE 1	→		
Communic	ation diversion is performed		
	INVITE 2		→ INVITE
	MESSAGE		
	200 OK MESSAGE		
100 B: :	180 Ringing	←	← 180 Ringing
180 Ringing	← 180 Ringing 200 OK (INVITE)	~	← 200 OK (INVITE)
200 OK (INVITE)	200 OK (INVITE) ← 200 OK (INVITE)	•	200 OK (INVITE)
ACK	200 OK (IIVITE)		→ ACK
/ 1011	Communica	ation	2 /1011
BYE	→	-	→ BYE
200 OK (BYE)	←		← 200 OK (BYE)
The test case can not b	e tested with an ATS		,

SSS_XXSSCFU 10	NGN reference to:		
TSS reference:	TS 183 004 [17] clause 4.5.2.6.3 SIP-SIP-SIP/Supplementary_services/C		
Configuration	The user B is in network N2 and is provi		
Selection criteria:		ded with CFU	
Selection chiena.	Call forwarding by the network. Call forwarding upportitional augustications.	a a ret a d	
	Call forwarding uncontitional supply a user C is not work determined user.		
Test purpose:	user C is network determined use The served user subscribes to the CFU		tion of communication
rest purpose.	diversion to the diverting user using the		
	communication is requested.	WEGGAGE request when	a new oaigoing
	Ensure that a diverting user will be inforr has initiated a new outgoing communica		
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 pre	econdition.	
Comments:		OID#O	015.40
SIP#1	SUT	SIP#2 (served user)	SIP#3
INVITE 1	→		
Communi	cation diversion is performed	•	3
	INVITE 2		→ INVITE
400 Dia sia s	180 Ringing	←	← 180 Ringing
180 Ringing	← 180 Ringing 200 OK (INVITE)	←	← 200 OK (INVITE)
200 OK (INVITE) ACK	← 200 OK (INVITE)→		→ ACK
		→ INVITE 3→ MESSAGE200 OK MESSAGE	→ INVITE
		€ 180 Ringing	← 180 Ringing
		← 200 OK (INVITE)	← 200 OK (INVITE)
		→ ACK	→ ACK
The test case can not	be tested with an ATS	7 AUR	AOR

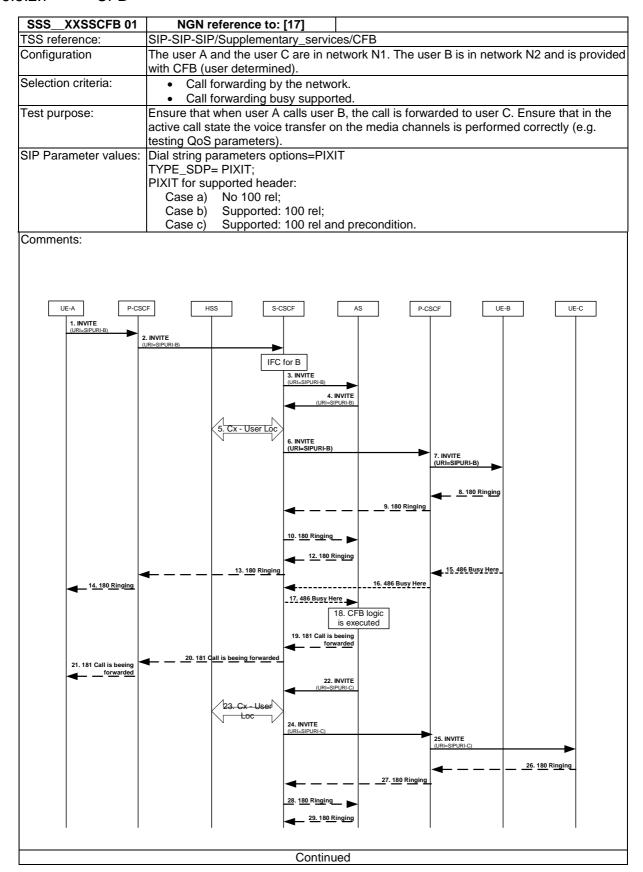
SSS_XXSSCFU 11	NGN reference to: TS 183 004 [17] clause 4.5.2.6.3			
TSS reference:	SIP-SIP-SIP/Supplementary_services	/CFU		
Configuration	The user B is in network N2 and is pro			
Selection criteria:	Call forwarding by the network.			
Coloculon ontona.		Call forwarding uncontitional supported.		
	user C is network determined user.			
Test purpose:	The served user subscribes to the CF		ion of communication diversion	
rest purpose.	to the diverting user using a voicemail		on or communication arvordion	
	Ensure that when communication dive			
	mail system request including the info	ormation where the call is div	rerted to.	
SIP Parameter	Dial string parameters options=PIXIT			
values:	TYPE_SDP= PIXIT;			
	PIXIT for supported header:			
	Case a) No 100 rel;			
	Case b) Supported: 100 rel;	100		
0 1	Case c) Supported: 100 rel and	precondition.		
Comments: SIP#1	SUT	SIP#2 (corred	SIP#3	
SIF#1	301	SIP#2 (served user)	317#3	
INVITE 1	→	usery		
	nication diversion is performed			
	INVITE	2 →	→ INVITE	
	INVITE	2 → Voicemail or Message mail system	→ INVITE	
400 Dinging	180 Ringir	→ Voicemail or Message mail system	→ INVITE ← 180 Ringing	
180 Ringing		→ Voicemail or Message mail system		
200 OK (INVITE)	180 Ringirg ← 180 Ringing 200 OK (INVITI) ← 200 OK (INVITE)	→ Voicemail or Message mail system	← 180 Ringing← 200 OK (INVITE)	
	180 Ringing ← 180 Ringing 200 OK (INVITE) →	→ Voicemail or Message mail system The system ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ←	← 180 Ringing	
200 OK (INVITE) ACK	180 Ringirg	→ Voicemail or Message mail system The system ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ←	← 180 Ringing← 200 OK (INVITE)→ ACK	
200 OK (INVITE) ACK BYE	180 Ringing	→ Voicemail or Message mail system The system ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ←	 ← 180 Ringing ← 200 OK (INVITE) → ACK → BYE 	
200 OK (INVITE) ACK BYE 200 OK (BYE)	180 Ringirg	→ Voicemail or Message mail system The system ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ←	← 180 Ringing← 200 OK (INVITE)→ ACK	

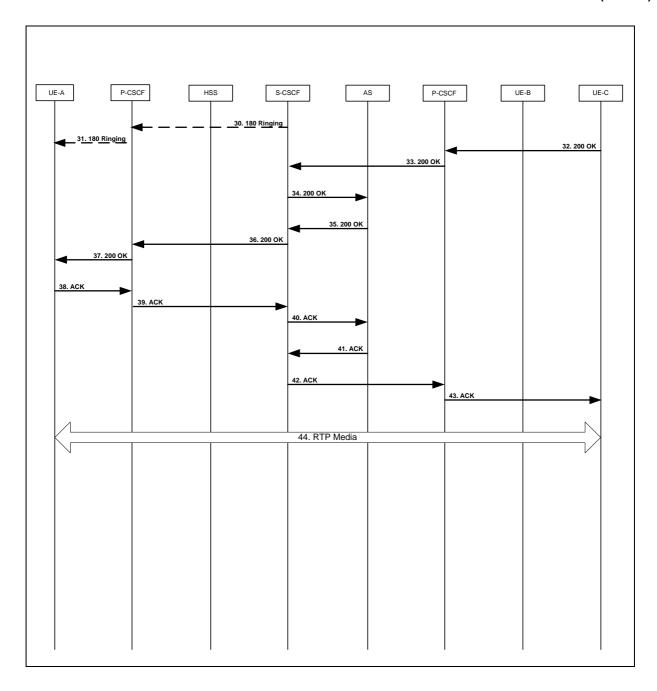
SSS_XXSSCFU 12			
	TS 183 004 [17] clause 4.5.2.6.3		
TSS reference:	SIP-SIP-SIP/Supplementary_services/		
Configuration	The user B is in network N2 and is pro-	vided with CFU.	
Selection criteria:	 Call forwarding by the network. 		
	 Call forwarding uncontitional su 		
	user C is network determined user		
Test purpose:	The served user subscribes to the CFU		
	to the diverting user using a voicemail	or mail system requested by	a timer.
	Ensure that when communication diver		
	mail system request including the infor		
	Message mail system request that is b	e sent due to a timer value t	hat can be provided by the
OID Demonster	user.		
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;		
values.	PIXIT for supported header:		
	Case a) No 100 rel;		
	Case b) Supported: 100 rel;		
	Case c) Supported: 100 rel and p	recondition.	
Comments:	, , , , , , , , , , , , , , , , , , , ,		
SIP#1	SUT	SIP#2 (served	SIP#3
		user)	
INVITE 1	→		
Commu	unication diversion is performed		
	INVIT	E2 →	→ INVITE
		→ Voicemail or Message mail system	
	180 Ring	ing 🗲	← 180 Ringing
180 Ringing	← 180 Ringing		
	200 OK (INVI	TE) ←	← 200 OK (INVITE)
200 OK (INVITE)	← 200 OK (INVITE)		3
ACK	→		→ ACK
BYE	Commun	ICATION	→ BYE
200 OK (BYE)	→		→ BYE ← 200 OK (BYE)
	ot be tested with an ATS		€ 200 ON (BTE)
The test case call if	ot be tested with an ATO		

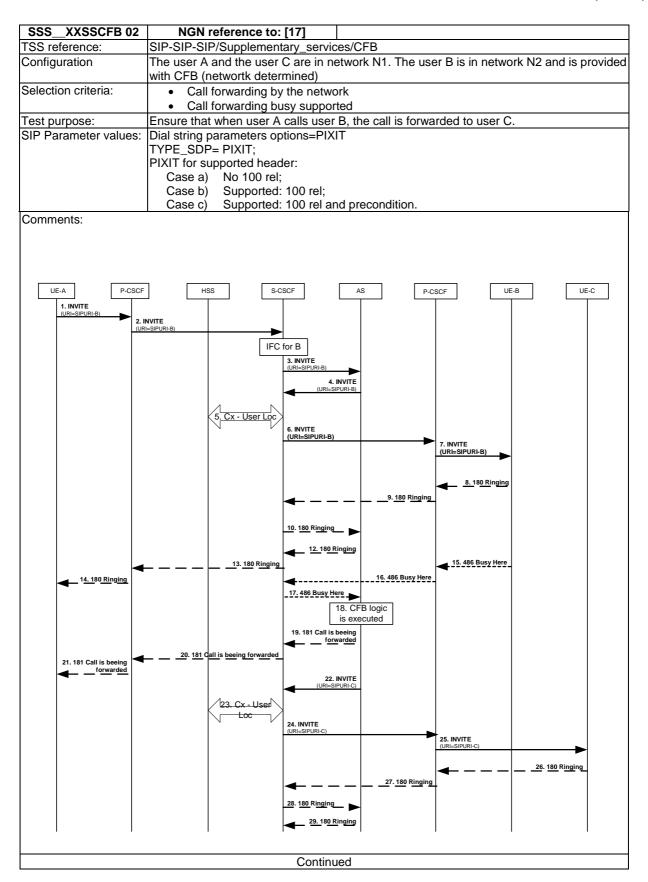
SSS_XXSSCFU 13				
T00 (TS 183 004 [17] clause 4.5.2.6.3	0511		
TSS reference:	SIP-SIP-SIP/Supplementary_services/		::. 0511	
Configuration	The user B is in network N2 and is pro	vided w	ith CFU.	
Selection criteria:	Call forwarding by the network.			
Ì	Call forwarding uncontitional su			
- .	user C is network determined user C is n			" · " · "
Test purpose:	The served user subscribes to the CFU			
l	communication diversion to the diverting	ng user	using a voicemail or	maii system.
l	Ensure that when communication diver			
Ì	mail system request including the infor			
Ì	be informed periodically with a Voicem	ail or M	essage mail system	request the information where
212.2	the call is diverted to.			
SIP Parameter	Dial string parameters options=PIXIT			
values:	TYPE_SDP= PIXIT;			
Ì	PIXIT for supported header: Case a) No 100 rel;			
Ì	Case b) Supported: 100 rel;			
Ì	Case c) Supported: 100 rel and p	recond	ition	
Comments:	Cacco, Capponea. 100 for ana p	71000110		
SIP#1	SUT		SIP#2 (served user)	SIP#3
INVITE 1	→		,	
Commu	nication diversion is performed			
Ì	INVITE	2 →		→ INVITE
Ì		→	Voicemail or	
Ì			Message mail	
l			system	
l	180 Ringir	ng 🗲		← 180 Ringing
180 Ringing	← 180 Ringing			
OOO OK (INI) (ITE)	200 OK (INVITE)	E) ←		← 200 OK (INVITE)
200 OK (INVITE) ACK	← 200 OK (INVITE)			→ ACK
AOIC	•		Voicemail or	AON
			Message mail system	
ı	Communi	ication		
	→			→ BYE
BYE	•			
200 OK (BYE)	to be tested with an ATS			← 200 OK (BYE)

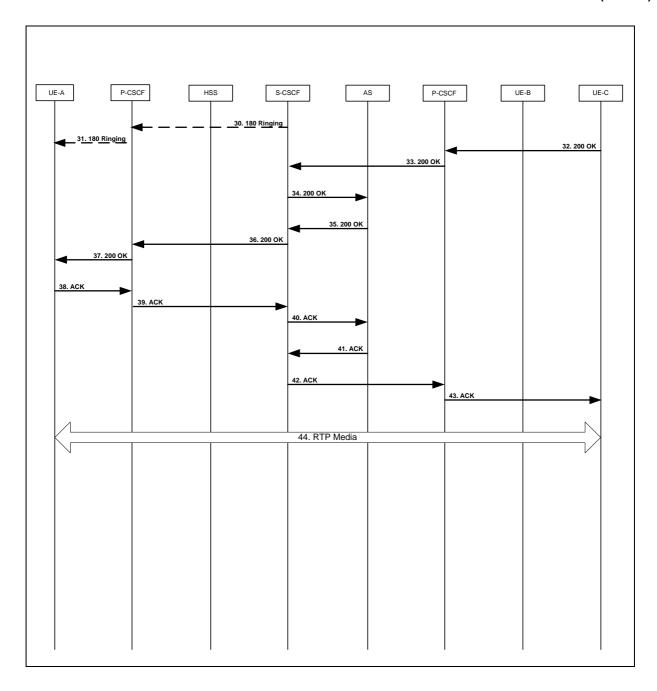
SSS_XXSSCFU 14	NGN reference to: TS 183 004 [17] clause 4.5.2.6.3		
TSS reference:	SIP-SIP/Supplementary_services	/CFU	
Configuration	The user B is in network N2 and is pro		
Selection criteria:	Call forwarding by the network		
Colodion ontona.	Call forwarding uncontitional si		
	user C is network determined to		
Test purpose:	The served user subscribes to the CF	U simulation service: Indicatio	n of a new communication
	diversion to the diverting user using a		
	Ensure that a diverting user will be inf	ormed with a Voicemail or Mes	ssage mail system request
	after the diverting user has initiated a		
	call is diverted to.		
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		
Commonto	Case c) Supported: 100 rel and	precondition	
Comments: SIP#1	SUT	SIP#2 (served user)	SIP#3
INVITE 1	→	userj	
	ation diversion is performed		
	INVITE 2	→ →	INVITE
		→ Voicemail or Message mail system	
180 Ringing	180 Ringing ← 180 Ringing	←	- 180 Ringing
200 OK (INVITE)	200 OK (INVITE) • 200 OK (INVITE)	←	- 200 OK (INVITE)
ACK `	→ Communic		ACK
BYE	→	-	▶ BYE
200 OK (BYE)	←	*	- 200 OK (BYE)
		→ INVITE 3 → Voicemail or Message mail system	NVITE
		← 180 Ringing 180 Ringing	- 180 Ringing
		← 200 OK (INVITE) ←	- 200 OK (INVITE)
The test case can not	be tested with an ATS		

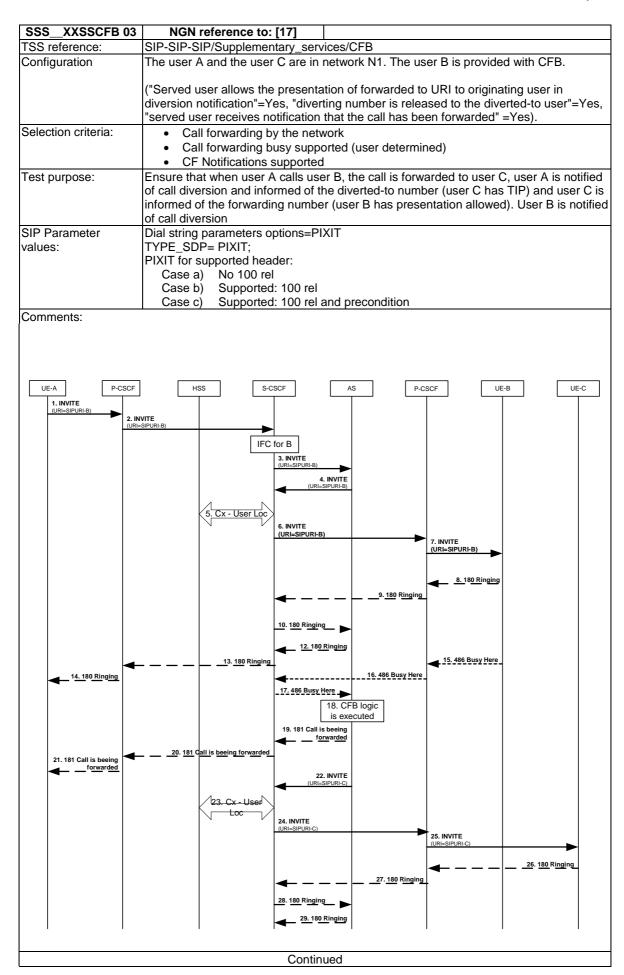
6.5.2.7 CFB

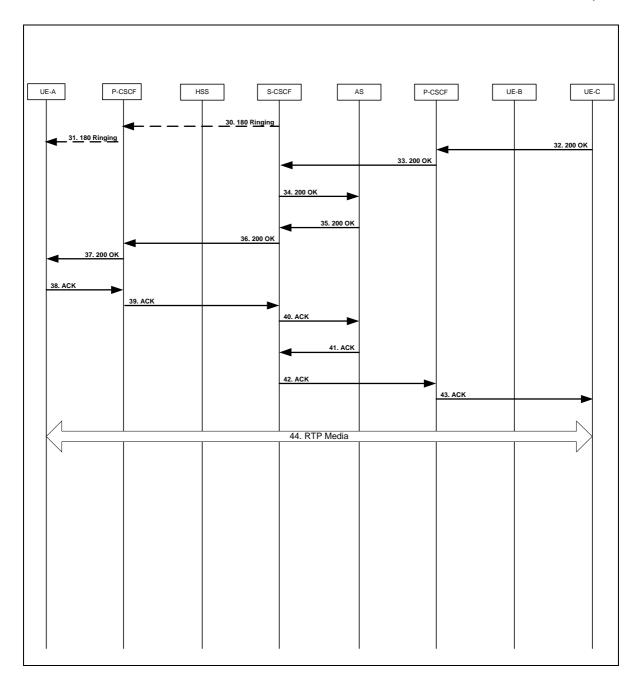


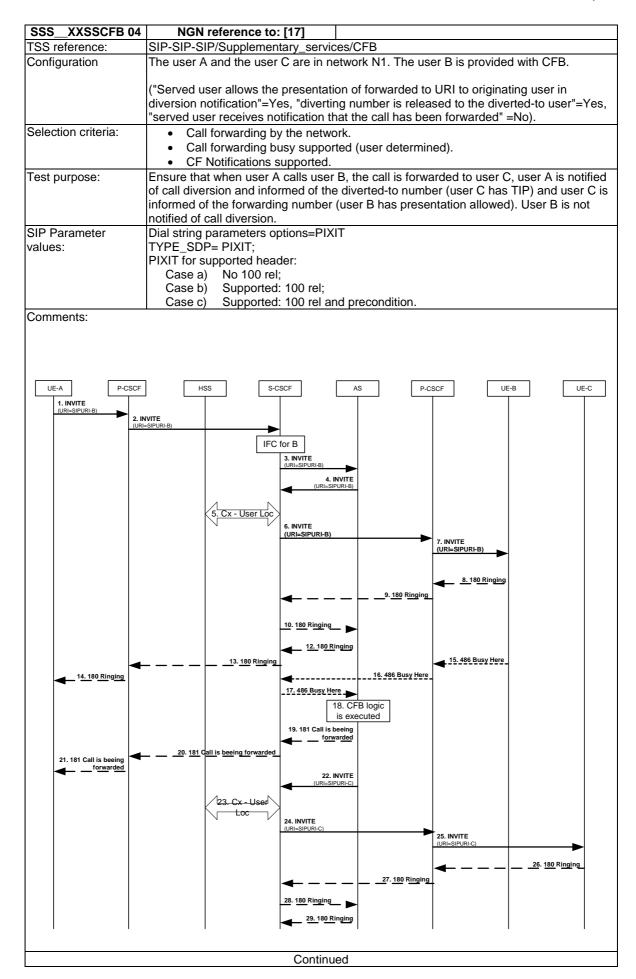


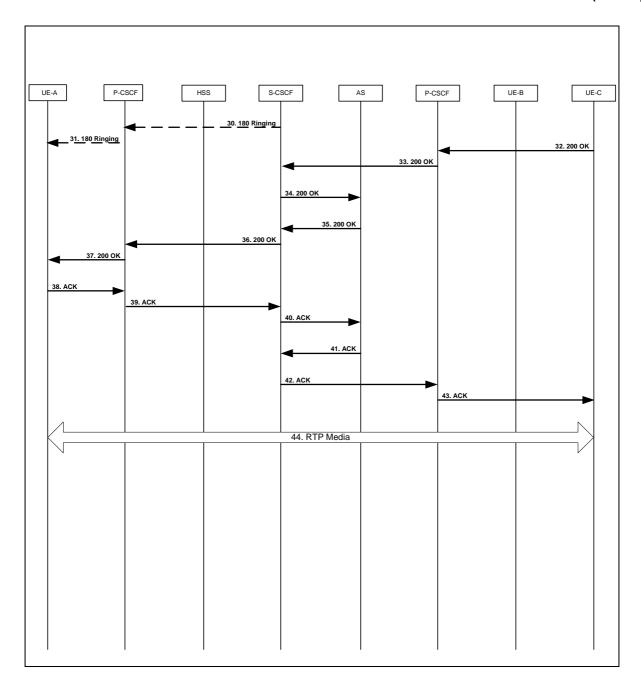


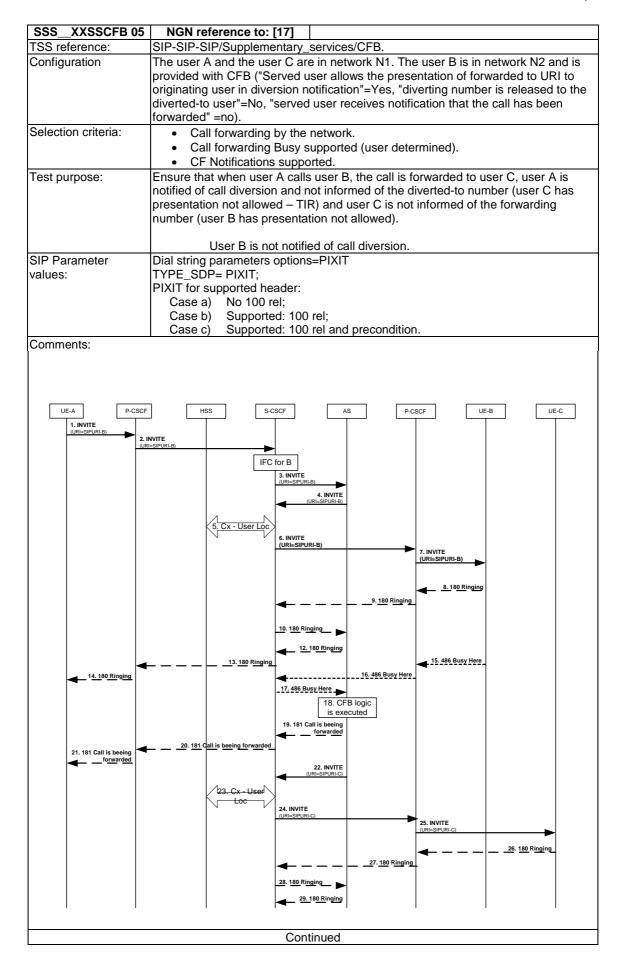


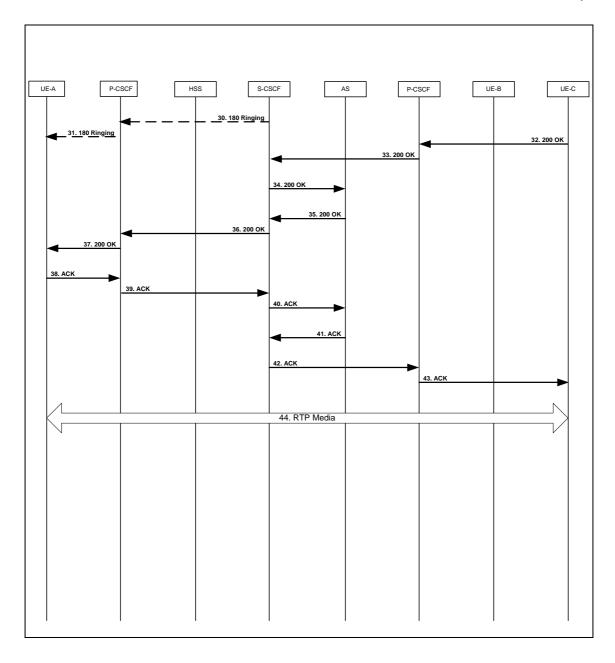


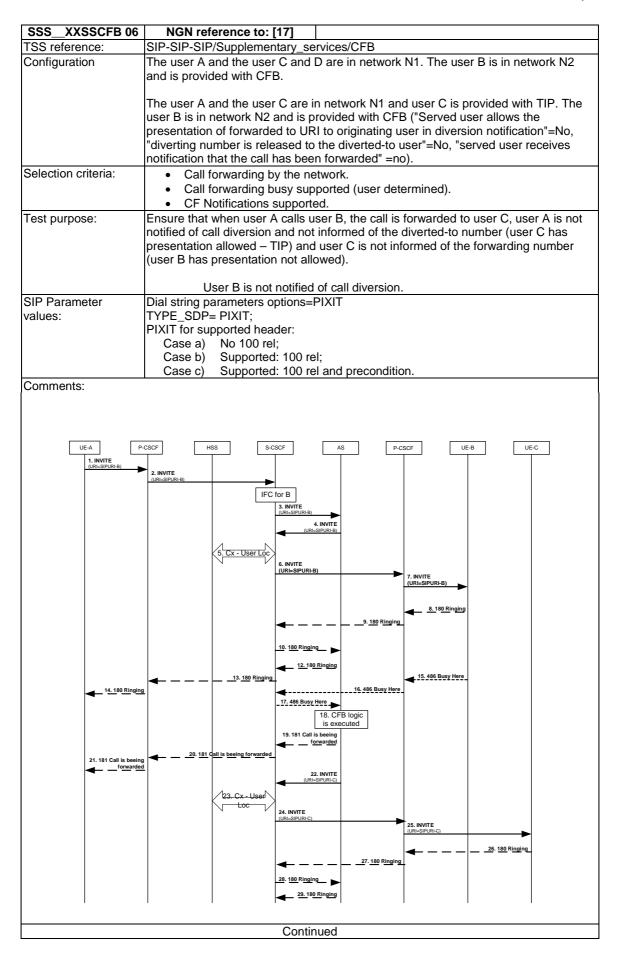


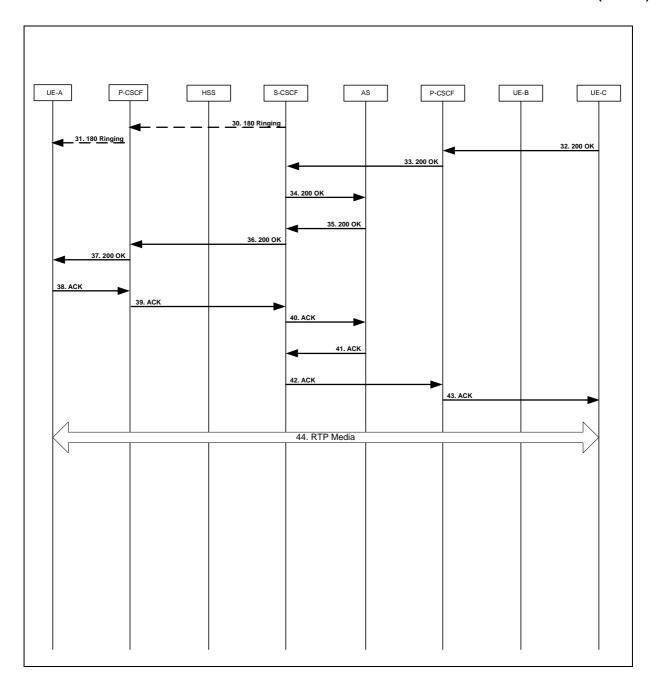


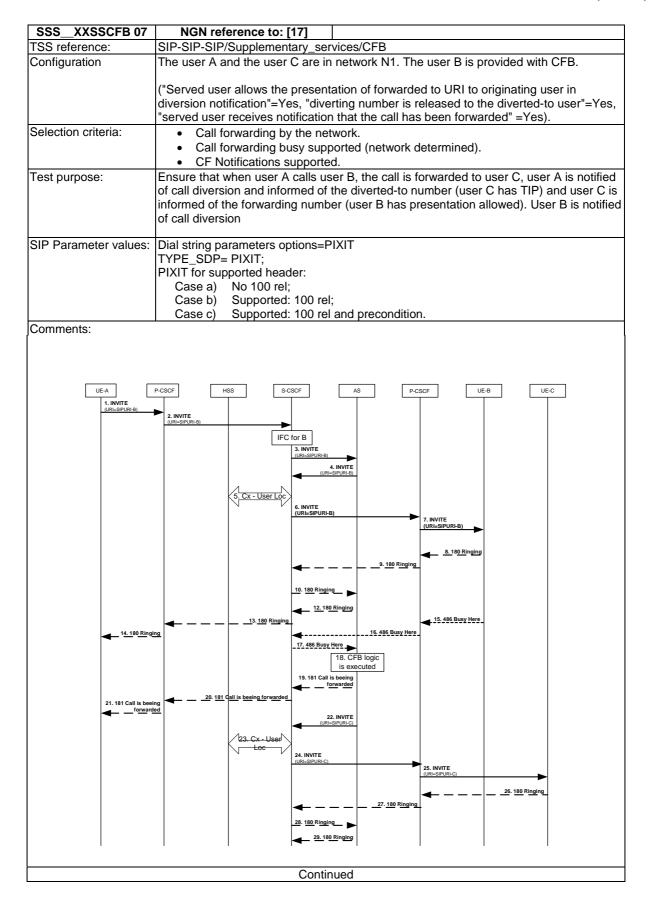


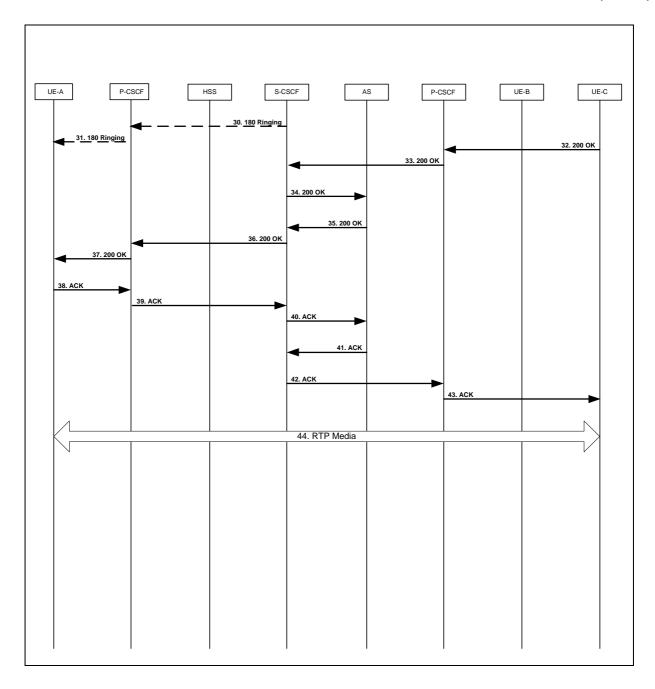


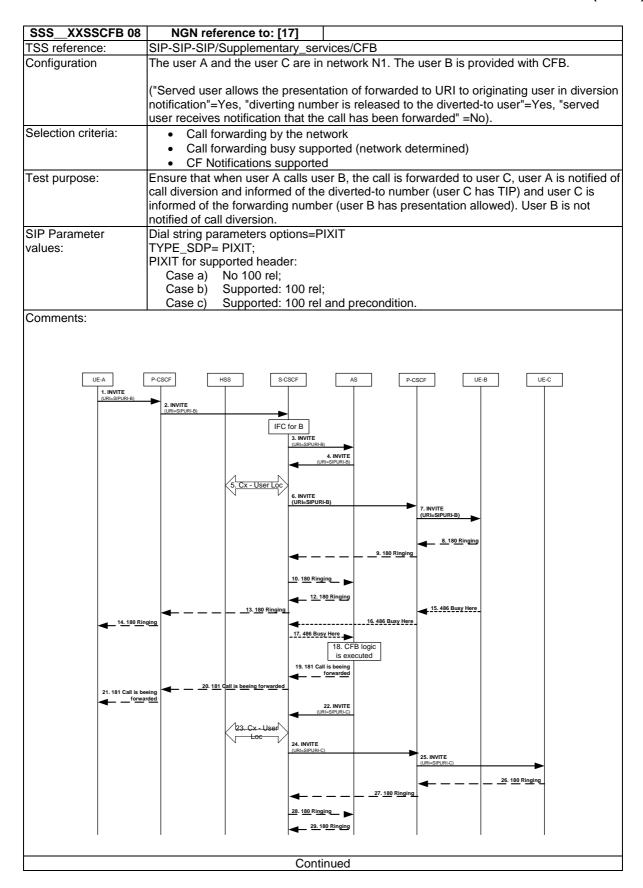


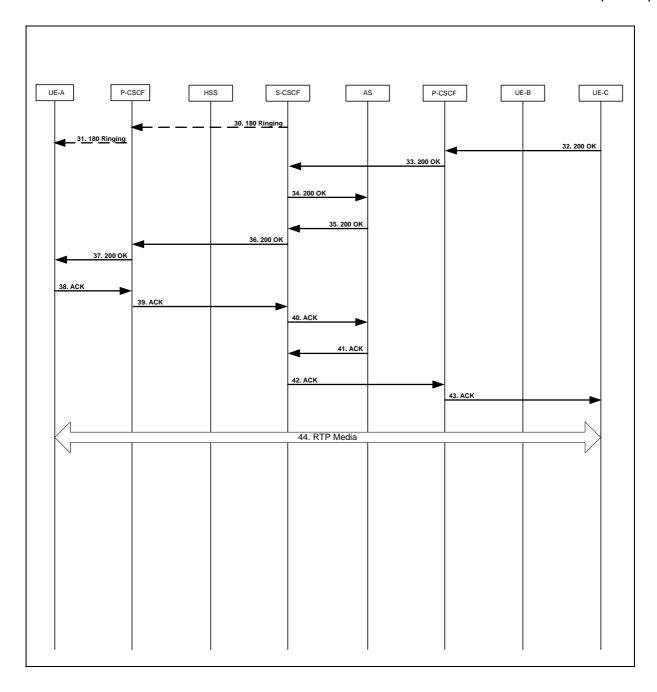


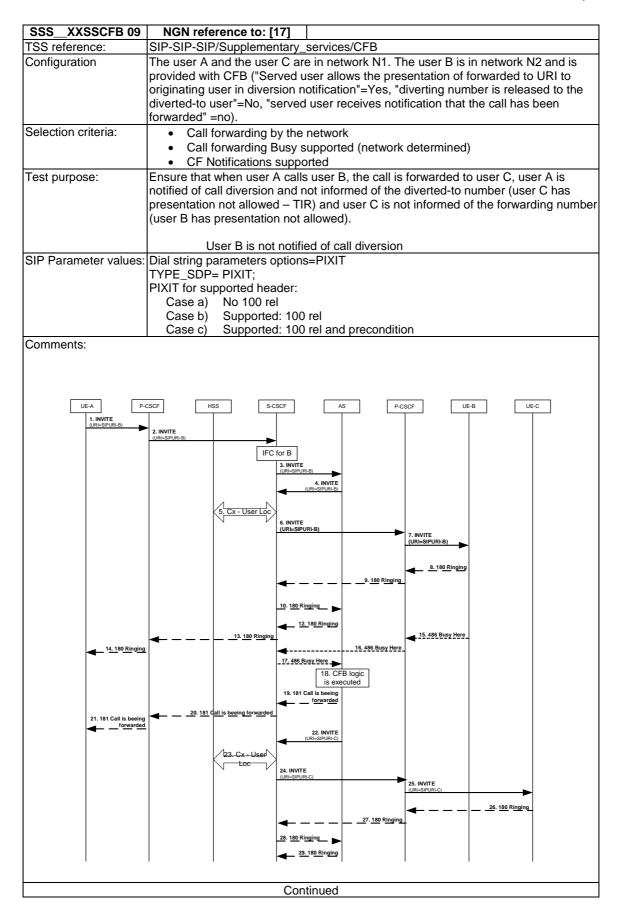


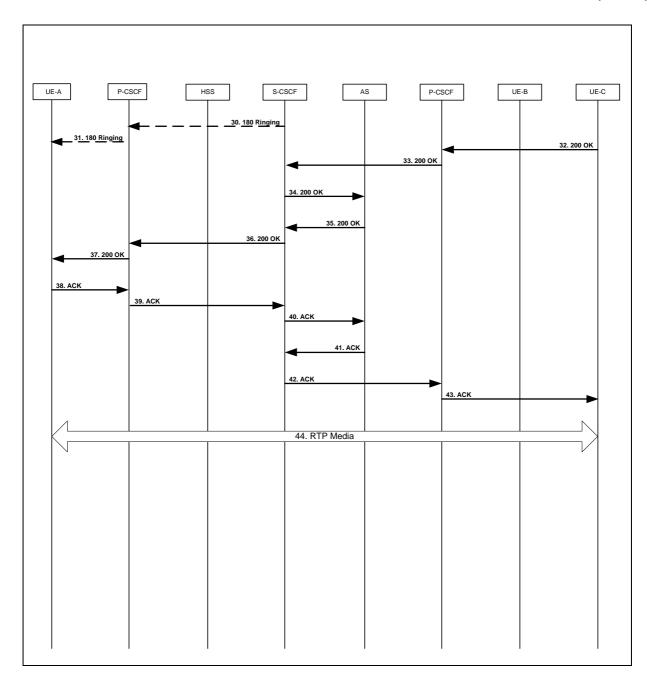




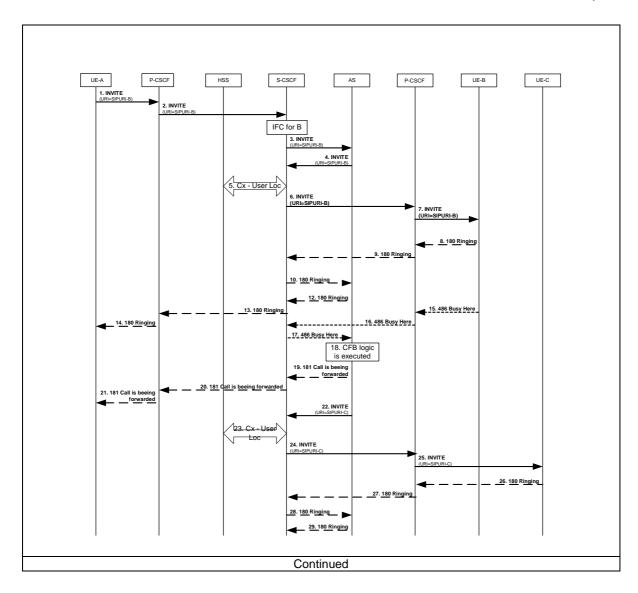


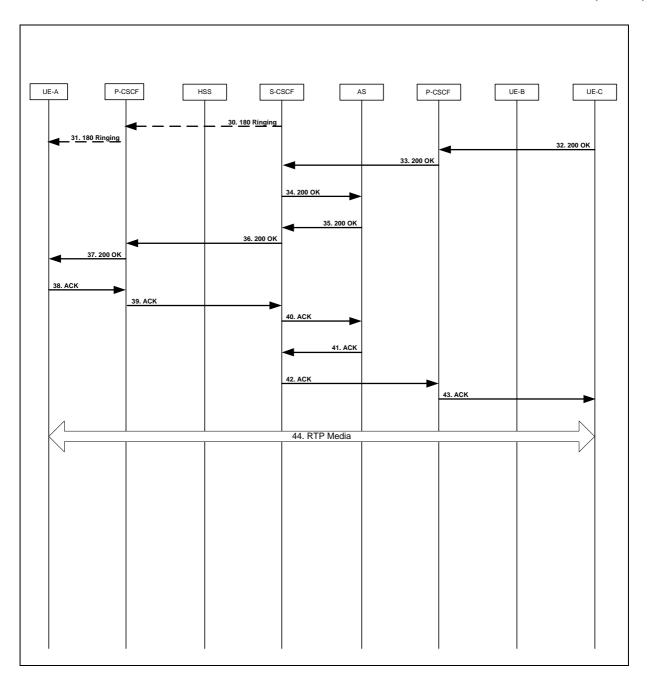






SSS_XXSSCFB 10	NGN reference to: [17]
TSS reference:	SIP-SIP/Supplementary_services/CFB
Configuration	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFB
	The user A and the user C are in network N1 and user C is provided with TIP. The user B is in network N2 and is provided with CFB ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=No, "diverting number is released to the diverted-to user"=No, "served user receives notification that the call has been forwarded" =no).
Selection criteria:	 Call forwarding by the network Call forwarding busy supported (network determined) CF Notifications supported
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is not notified of call diversion and not informed of the diverted-to number (user C has presentation allowed – TIP) and user C is not informed of the forwarding number (user B has presentation not allowed).
	User B is not notified of call diversion
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;
	PIXIT for supported header: Case a) No 100 rel
	Case a) No 100 fel
	Case c) Supported: 100 rel and precondition
Comments:	, ,
	Continued





SSS_XXSSCFB 15	NGN reference to: [17]					
TSS reference:	SIP-SIP/Supplementary_services/CFB					
Configuration	The user B is in network N2 and is provided with CFB					
Selection criteria:	Call forwarding by the network.					
	Call forwarding busy supported.					
	user C is user determined user busy.					
Test purpose:	To verify that a call is released correctly if CFB was not successful.					
	User A calls user B, the call is forwarded to user C who is user determined					
	user busy.					
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:						

SSS_XXSSCFB 16	NGN reference to: [17]				
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFB				
Configuration	The user B is in network N2 and is provided with CFB				
Selection criteria:	Call forwarding by the network.				
	Call forwarding busy supported.				
	user C is network determined user busy.				
Test purpose:	To verify that a call is released correctly if CFB was not successful.				
	User A calls user B, the call is forwarded to user C who is network determined user busy.				
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:					

SSS_XXSSCFB 17		Reference TS 183 004 [17]					
		clause 4.5.2.6.3					
TSS reference:		P-SIP/Supplementary_	services/C	FB			
Configuration		er B is in network N2 a					
Selection criteria:		Call forwarding by the					
		Call forwarding busy s					
		user C is user determi		usy	<i>/</i> .		
Test purpose:		rved user subscribes t				igina	iting user is
	notified	d. Additionaly an annou	incement is	s pl	ayed.		
	origina be sen Additio in orde	Communication Divers ting user is supported towards the originational the AS initiates and to inform about the diverse and a site of the communication of th	then a 181 ig user. announcei iversion. A	(Ca mer	all Is Being Forwar nt to be included to ouncements shall b	ded war) response shall ds the calling user
SIP Parameter values:		lures as are described		028	3 [18].		
SIF Farailleter values.		SDP= PIXIT;	S=FIAII				
		for supported header:					
	Cas	se a) No 100 rel					
		se b) Supported: 100					
		se c) Supported: 100	rel and pr	eco	ndition		
		eader values: all is Being Forwarded					
Comments:	101 0	all is being Forwarded					
Commonto.							
SIP#1		SUT			CID#0 /		
_					SIP#2 (served user)		SIP#3
INVITE 1	→	INVITE		_	user)		SIP#3
_	→	INVITE	INVITE		user)		SIP#3
_	→	INVITE		(user) INVITE 486 Busy Here		SIP#3
INVITE 1 181 Call is Being	→	INVITE	usy Here ACK	(user) INVITE 486 Busy Here		SIP#3
INVITE 1		INVITE 486 B	usy Here ACK Forwarded	← →	user) INVITE 486 Busy Here	→	
INVITE 1 181 Call is Being		INVITE 486 Bit 181 Call is Being F	usy Here ACK Forwarded INVITE	← →	user) INVITE 486 Busy Here		INVITE
INVITE 1 181 Call is Being	←	INVITE 486 Bit 181 Call is Being F	usy Here ACK Forwarded	← →	user) INVITE 486 Busy Here		
INVITE 1 181 Call is Being Forwarded 180 Ringing	+	181 Call is Being F 180 Ringing 200 OK	usy Here ACK Forwarded INVITE	← → ←	user) INVITE 486 Busy Here	←	INVITE
INVITE 1 181 Call is Being Forwarded 180 Ringing 200 OK (INVITE)	+ + +	INVITE 486 Bit 181 Call is Being Financial 180 Ringing	usy Here ACK Forwarded INVITE Ringing	← → ←	user) INVITE 486 Busy Here	+	INVITE 180 Ringing 200 OK (INVITE)
INVITE 1 181 Call is Being Forwarded 180 Ringing	+	181 Call is Being I 180 Ringing 200 OK 200 OK (INVITE)	usy Here ACK Forwarded INVITE 9 Ringing (INVITE)	← → ←	user) INVITE 486 Busy Here	+	INVITE 180 Ringing 200 OK
181 Call is Being Forwarded 180 Ringing 200 OK (INVITE) ACK	÷ ÷	181 Call is Being I 180 Ringing 200 OK 200 OK (INVITE)	usy Here ACK Forwarded INVITE Ringing	← → ←	user) INVITE 486 Busy Here	← ←	INVITE 180 Ringing 200 OK (INVITE) ACK
INVITE 1 181 Call is Being Forwarded 180 Ringing 200 OK (INVITE)	+ + +	181 Call is Being I 180 Ringing 200 OK 200 OK (INVITE)	usy Here ACK Forwarded INVITE 9 Ringing (INVITE)	← → ←	user) INVITE 486 Busy Here	←←→→	INVITE 180 Ringing 200 OK (INVITE)

SSS_XXSSCFB 18	Reference TS 183 004 [17] clause 4.5.2.6.3		
TSS reference:	SIP-SIP-SIP/Supplementary_services/0	L CFR	
Configuration	The user B is in network N2 and is prov		
Selection criteria:	Call forwarding by the network. Call forwarding busy supported. user C is user determined user by		
Test purpose:	The served user subscribes to the CFB communication diversion to the diverting triggered by a timer. Ensure that when the diverting user is request including the information where request that is be sent due to an timer v	simulation service; Indicating user using the MESSAGE egistering the AS sends a Nathe call is diverted to. The	request MESSAGE MESSAGE
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 pr		y 4.10 door.
Comments:			
SIP#1	SUT	SIP#2	SIP#3
INVITE 1	486 Busy Here ACK INVITE 2	→ MESSAGE ← 200 OK	INVITE 180 Ringing
180 Ringing	← 180 Ringing 200 OK (INVITE)	← ←	200 OK (INVITE)
200 OK (INVITE) ACK	← 200 OK (INVITE) → Communication	→	ACK
BYE 200 OK (BYE)	→ ←	-	BYE 200 OK (BYE)

SSS_XXSSCFB 19		Reference				
		3 004 [17] clause 4.5.2.6.3				
TSS reference:		IP/Supplementary_services/0				
Configuration		is in network N2 and is prov	/ide	d with CFB		
Selection criteria:		I forwarding by the network.				
		I forwarding busy supported.				
_	• usei	r C is user determined user	ousy	/.		
Test purpose:	communica	d user subscribes to the CFB ation diversion to the divertin ing communication is reques	g us	ser using the MES		
	diverting us call is diver					
SIP Parameter		parameters options=PIXIT				
values:	TYPE_SDF	P= PIXII; upported header:				
	Case a)					
	Case b)	,				
	Case c)		reco	ondition		
Comments:						
SIP#1	5 11 11 1	SUT		SIP#2 (served user)		SIP#3
INVITE 1	→ INV	/ITE INVITE	_	INI\/ITE		
		486 Busy Here ACK	←			
		INVITE 2	→		→	INVITE
		MESSAGE 180 Ringing		MESSAGE	←	180 Ringing
		200 OK		200 OK		100 Kinging
180 Ringing	← 180	Ringing 200 OK (INVITE)	_		_	200 OK (INVITE)
200 OK (INVITE)	← 200	OK (INVITE)	•		_	200 OK (INVITE)
ACK	→	Communication			→	ACK
		INVITE 2	→	MESSACE	→	INVITE
		MESSAGE 180 Ringing	7	MESSAGE	←	180 Ringing
		roo ranging	_	200 OK (INVITE)		200 OK (INVITE)
			→	ACK	→	ACK
The test case can no	t be tested	with an ATS				

SSS_XXSSCFB 20	Reference				
	TS 183 004 [17] clause 4.5.2.6.3				
TSS reference:	SIP-SIP-SIP/Supplementary_services/0				
Configuration	The user B is in network N2 and is prov	/ided	with CFB		
Selection criteria:	 Call forwarding by the network. 				
	 Call forwarding busy supported. 				
	user C is user determined user				
Test purpose:	The served user subscribes to the CFB				
	communication diversion to the divertin	g use	er using a voicema	il or	mail system.
	Ensure that when communication diver				
	Message mail system request including	the	information where	the o	call is diverted to.
SIP Parameter	Dial string parameters options=PIXIT				
values:	TYPE_SDP= PIXIT;				
	PIXIT for supported header:				
	Case a) No 100 rel				
	Case b) Supported: 100 rel		!!#!		
0	Case c) Supported: 100 rel and p	recor	naition		
Comments:	CUT		SIP#2		CID#2
SIP#1	SUT		(served user)		SIP#3
INVITE 1	→ INVITE		(served user)		
IINVIIL I		4	INVITE		
			486 Busy Here		
			ACK		
	INVITE 2			→	INVITE
	Voicemail or Message mail		Voicemail or		
	system		Message mail		
	400 Dinging	_	system	_	400 Diamina
	180 Ringing	~		+	180 Ringing
180 Ringing	← 180 Ringing				
3 3	200 OK (INVITE)	←		←	200 OK
000 014 (INI) (ITE)	# 000 OK (INI) (ITE)				(INVITE)
200 OK (INVITE)	← 200 OK (INVITE)				4.017
ACK	Communication			→	ACK
BYE	Communication →			→	BYF
200 OK (BYE)	7 4			→	200 OK (BYE)
	t be tested with an ATS			•	200 ON (BTE)
The test case can no	t Do tooled With an ATO				

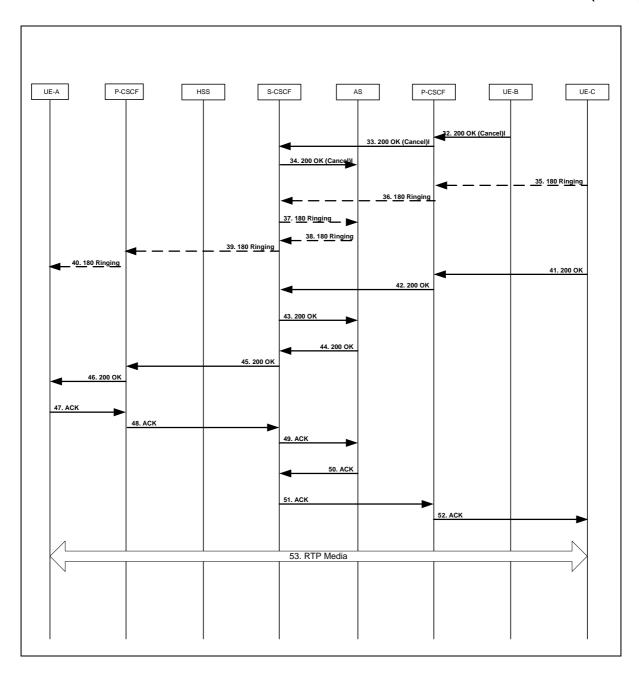
SSS_XXSSCFB 21	Reference	
	TS 183 004 [17] clause 4.5.2.6.3	
TSS reference:	SIP-SIP-SIP/Supplementary_services/	
Configuration	The user B is in network N2 and is prov	vided with CFB
Selection criteria:	 Call forwarding by the network. 	
	 Call forwarding busy supported. 	
	user C is user determined user	
Test purpose:	The served user subscribes to the CFE communication diversion to the divertin requested by a timer.	3 simulation service; Indication of ng user using a voicemail or mail system
	Message mail system request including	rsion is performed, the AS sends a Voicemail or g the information where the call is diverted to. n shall be sent due to a timer value that can be
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 p	precondition
Comments:	т толго од тограниот по	
SIP#1	SUT	SIP#2 SIP#3 (served user)
INVITE 1	486 Busy Here	→ Voicemail or Message mail system
180 Ringing 200 OK (INVITE)	 ← 180 Ringing 200 OK (INVITE) ← 200 OK (INVITE) 	← 200 OK (INVITE)
ACK (INVITE)	Communication	→ ACK
BYE	→ Communication	→ BYE
200 OK (BYE)	← t be tested with an ATS	€ 200 OK (BYE)

SSS_XXSSCFB 22	Reference						
	TS 183 004 [17] clause 4.5.2.6.3						
TSS reference:		SIP-SIP-SIP/Supplementary_services/CFB					
Configuration	The	The user B is in network N2 and is provided with CFB					
Selection criteria:		Call forwarding by the network.					
		Call forwarding busy supported.					
			C is user determined u				
Test purpose:	The	e served ι	user subscribes to the	CFB	simulation service	e; pe	riodically
			communication diversi	on to	o the diverting use	r usi	ng a voicemail or
	ma	il system.					
					-::	41	A O
			when communication o				
	001	Lie diverte	Message mail system ed to. The diverting use	requ	uest including the	odior	mation where the
			mail system with the in				
SIP Parameter values:			arameters options=PIX		iation where the c	all 13	diverted to.
on Tarameter values.		PE_SDP=					
			oported header:				
		Case a)	No 100 rel				
		Case b)	Supported: 100 rel				
		Case c)	Supported: 100 rel ar	nd pi	recondition		
Comments:							
SIP#1			SUT		SIP#2		SIP#3
INDUTE 4		IN 1) //TE			(served user)		
INVITE 1	→	INVITE	INIVITE	_	INVITE		
			486 Busy Here				
					ACK		
			INVITE 2		7.010	→	INVITE
		Voicen	nail or Message mail		Voicemail or	_	
			system		Message mail		
			•		system		
			180 Ringing	←		←	180 Ringing
180 Ringing	←	180 Ring		_		_	
			200 OK (INVITE)	←		←	200 OK
OOO OK (INI) (ITE)	,	000 014	/INI\ /ITE\				(INVITE)
200 OK (INVITE) ACK	7	200 OK	(INVITE)			→	ACK
ACK	7	Voicem	ail or Message mail	→	Voicemail or	7	ACK
		system	an or message man	-	Message mail		
		2,0.0			system		
			Communication		- ,		
BYE	→					→	BYE
200 OK (BYE)	←					←	200 OK (BYE)
The test case can not be	tes	ted with a	an ATS				

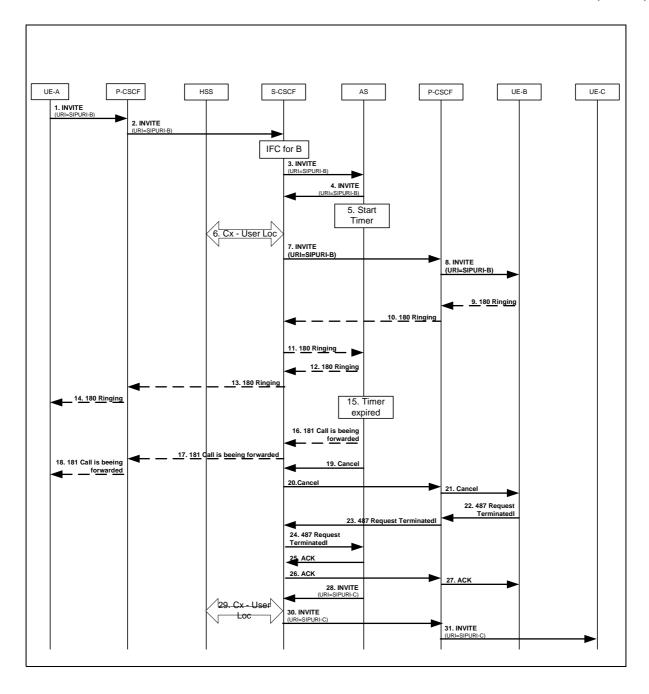
SSS XXSSCFB 23	Reference				1	
	TS 183 004 [17] clause 4.5.2.	6.3				
TSS reference:	SIP-SIP-SIP/Supplementary_serv		CFB			
Configuration	The user B is in network N2 and i					
Selection criteria:	Call forwarding by the net	vork.				
	Call forwarding busy supported.					
	user C is user determined					
Test purpose:	The served user subscribes to the			ce: Ir	ndication of a new	
	communication diversion to the di					
			0		,	
	Ensure that a diverting user will b	e info	rmed with Messa	ige n	nail system request	
	after the diverting user has initiate	ed a r	new outgoing com	ımur	nication the	
	information where the call is diver	ted to).			
SIP Parameter values:	Dial string parameters options=PI	XIT				
	TYPE_SDP= PIXIT;					
	PIXIT for supported header:					
	Case a) No 100 rel					
	Case b) Supported: 100 rel		11.1			
	Case c) Supported: 100 rel	and p	recondition			
Comments:	CUT		CID#0		CID#2	
SIP#1	SUT		SIP#2		SIP#3	
INVITE 1	→ INVITE		(served user)			
INVITE I	INVITE	4	INI\/ITE			
	486 Busy Here					
	ACK					
	Voicemail or Message mail		Voicemail or			
	system		Message mail			
	•		system			
	INVITE 2			→	INVITE	
	180 Ringing	←		←	180 Ringing	
180 Ringing	← 180 Ringing					
	200 OK (INVITE)	←		←	200 OK (INVITE)	
200 OK (INVITE)	← 200 OK (INVITE)					
ACK	→			→	ACK	
	Communicatio			_		
	INVITE 3				INVITE	
	180 Ringing	(400 D: :	←	180 Ringing	
			180 Ringing	_	200 OK (INI\/ITE)	
		-	200 OK	+	200 OK (INVITE)	
	Voicemail or Message mail		(INVITE) Voicemail or			
	system		Message mail			
	3,316111		system			
			ACK	→	ACK	
The test case can not be	e tested with an ATS	-		-		

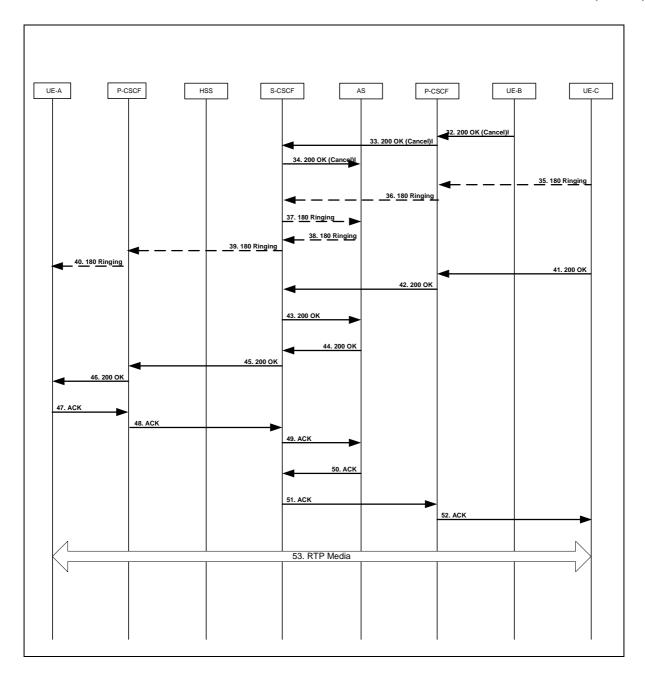
6.5.2.8 CFNR

SSS_XXSSCFNR01	NGN reference to: [17]
SS reference:	SIP-SIP/Supplementary_services/CFNR/
Selection criteria:	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR Served user communication retention on invocation of diversion (forwarding or deflection)= No [Clear call to the served user on invocation of call diversion].
est purpose:	Ensure that when user A calls user B, if unanswered, 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).
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:	Case of Capported. For the and precondition
UE-A 1. INVITE (URI=SIPURI-B) 2. INVI (URI=SII	
14. 180 Ringing	7. INVITE (URI=SIPURI-B) 8. INVITE (URI=SIPURI-B) 9. 180 Ringing 11. 180 Ringing 12. 180 Ringing
18. 181 Call is beeing forwarded	15. Timer expired 16. 181 Call is beeing forwarded 19. Cancel 21. Cancel
	23. 487 Request TerminatedI 24. 487 Request TerminatedI 25. ACK 26. ACK 28. INVITE (URI=SIPURI-C) 30. INVITE

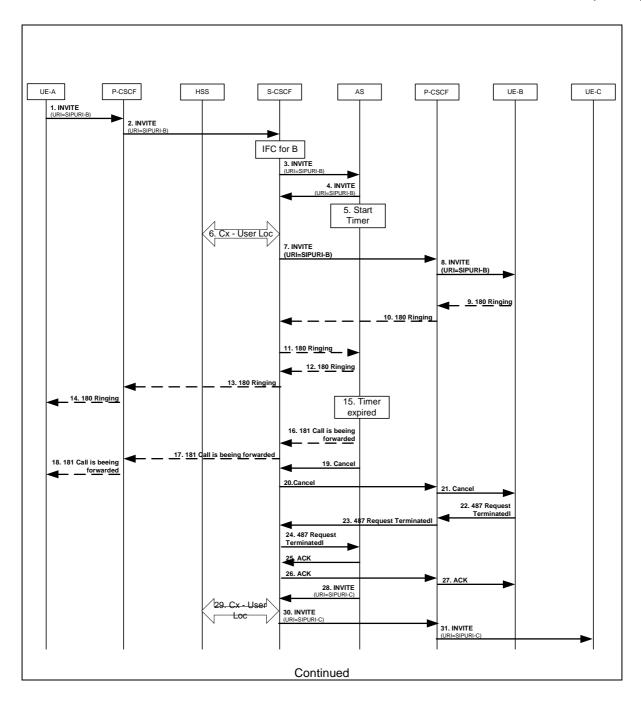


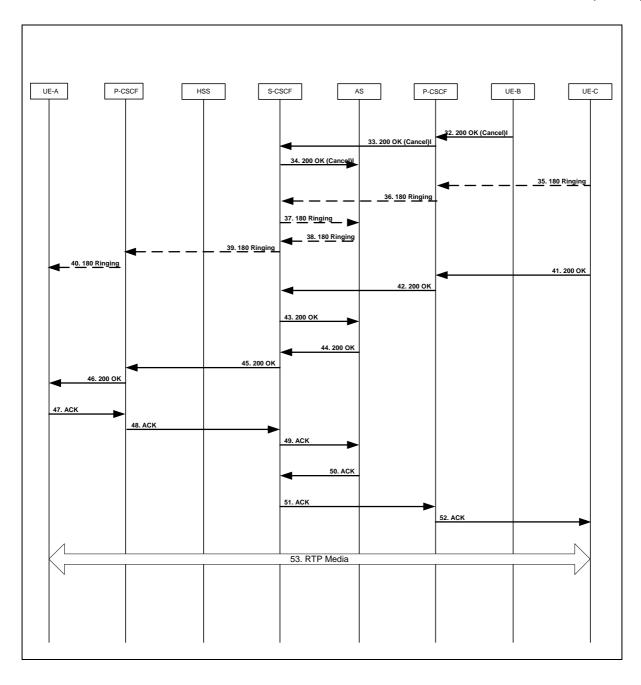
SSS_XXSSCFNR02	NGN reference to: [17]					
TSS reference:	SIP-SIP/Supplementary_services/CFNR/					
Configuration	The user A and the user C are in network N1. The user B is provided with CFNR.					
	("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting number is released to the diverted-to user"=Yes, "served user receives notification that the call has been forwarded" =Yes, Served user communication retention on invocation of diversion (forwarding or deflection)= No [Clear call to the served user on invocation of call diversion],					
Selection criteria:	 CFNR supported: CF Notifications supported. 					
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. User A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed). User B is notified of call diversion					
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:	1 25.5 3, 25.75					
	Continued					



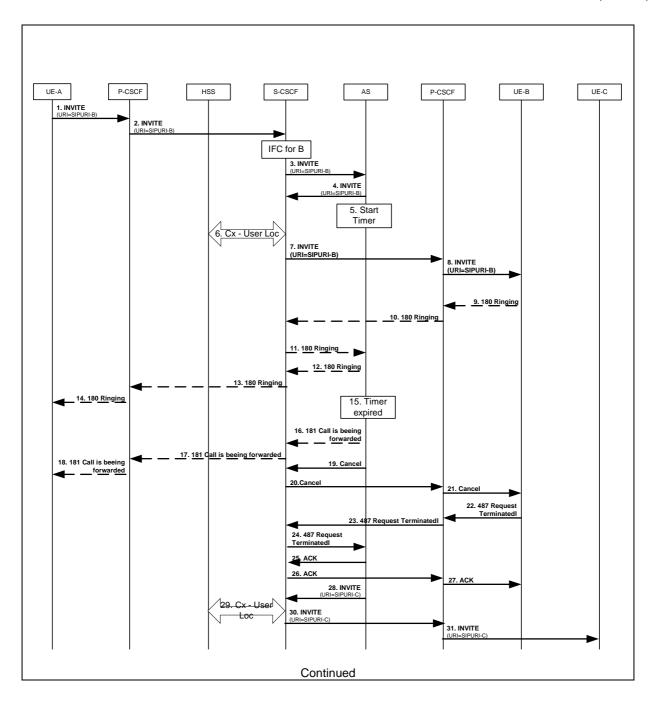


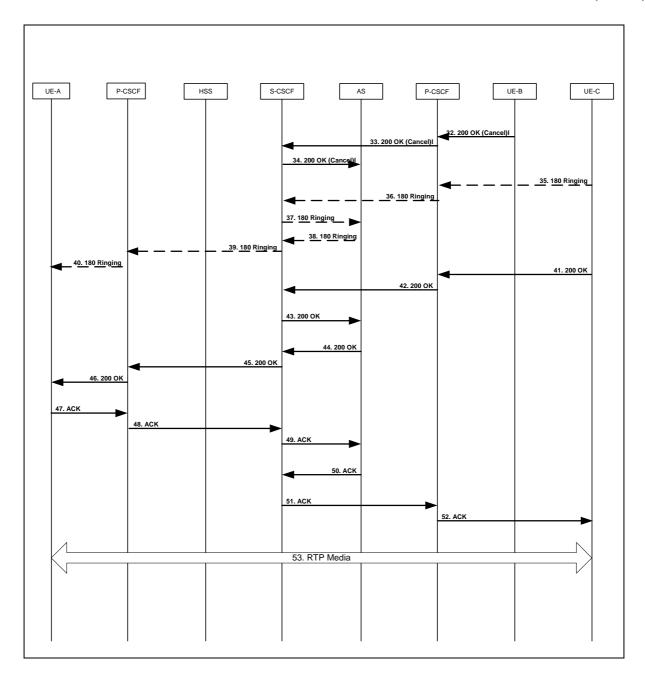
SSS_XXSSCFNR03	NGN reference to: [17]					
TSS reference:	SIP-SIP/Supplementary_services/CFNR/					
Configuration	The user A and the user C are in network N1. The user B is provided with CFNR.					
	("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting number is released to the diverted-to user"=Yes, "served user receives notification that the call has been forwarded" =No, Served user communication retention on invocation of diversion (forwarding or deflection)= No [Clear call to the served user on invocation of call diversion].					
Selection criteria:	CFNR supported: CF Notifications supported.					
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. User A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion					
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:	Continued:					



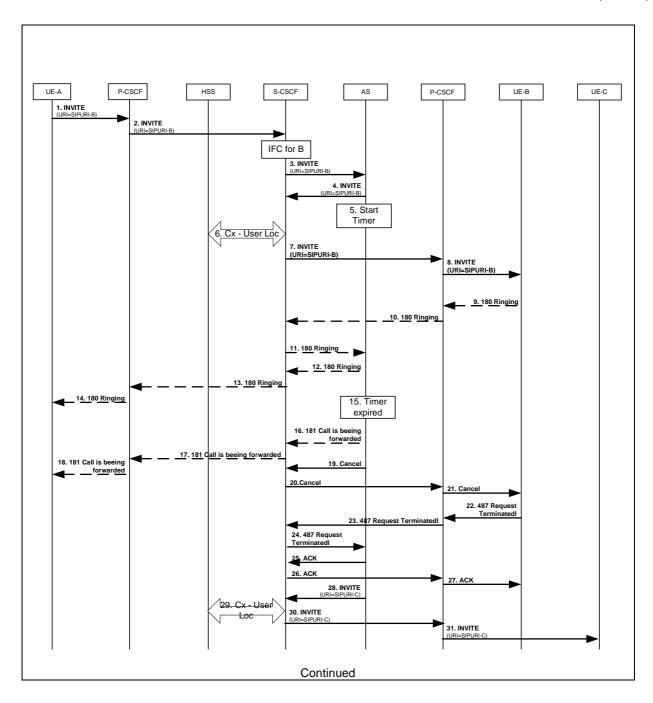


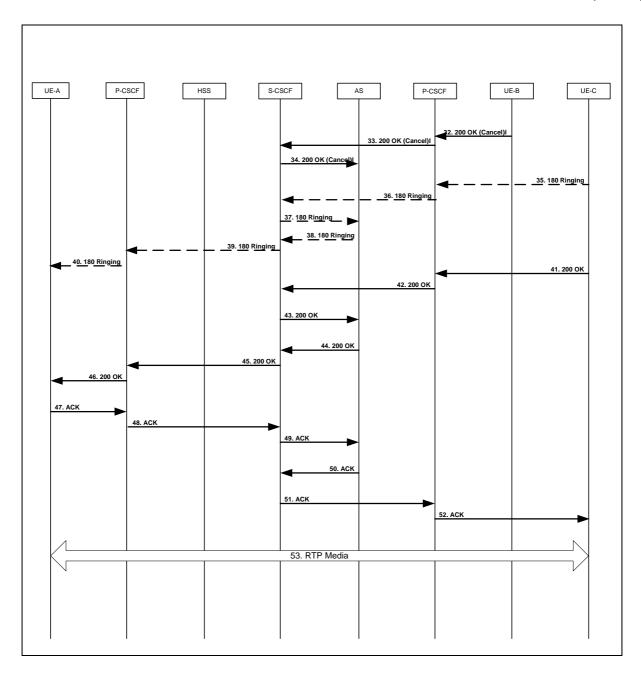
SSS_XXSSCFNR 04	NGN reference to: [17]	
TSS reference:	SIP-SIP/Supplementary_services/CFNR	
Configuration	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting number is released to the diverted-to user"=No, "served user receives notification that the call has been forwarded" =no) Served user communication retention on invocation of diversion (forwarding or deflection)= No.	
Selection criteria:	 Call forwarding by the network. Call forwarding not reply supported. CF Notifications supported. 	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed – TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion.	
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:	Continued:	



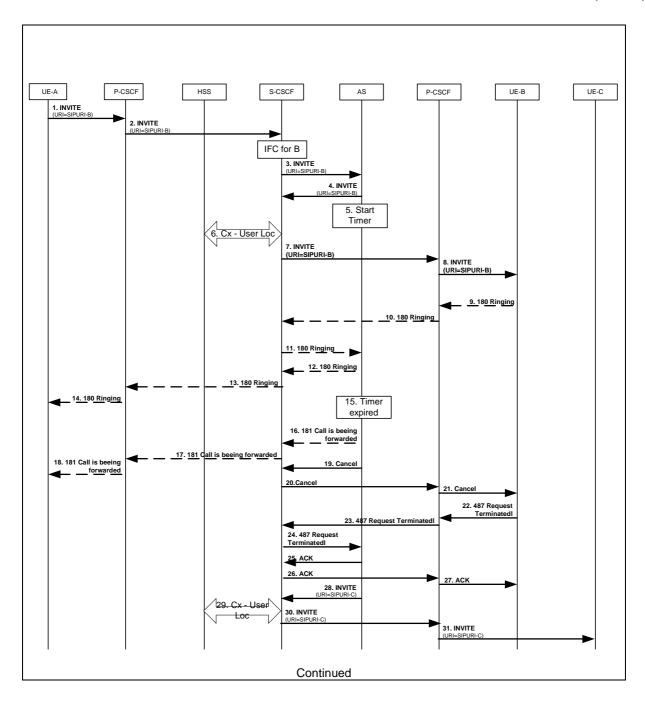


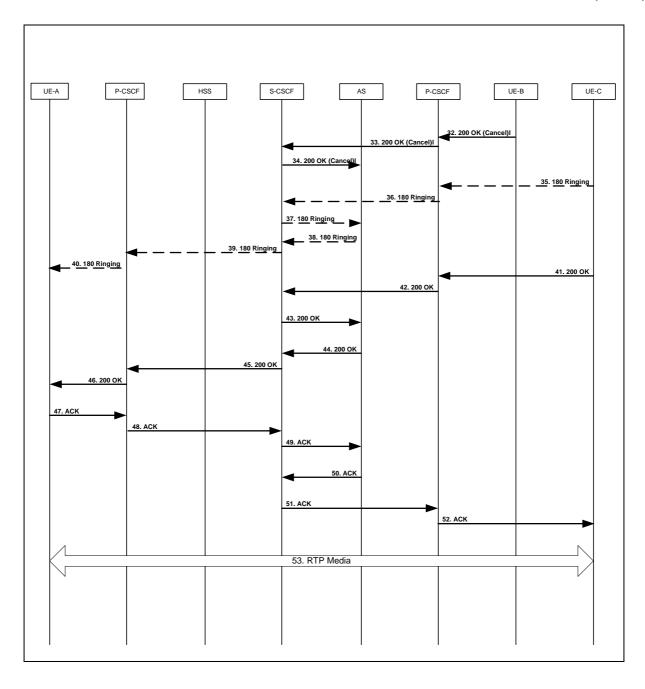
SSS_XXSSCFNR 05	NGN reference to: [17]		
TSS reference:	SIP-SIP/Supplementary_services/CFNR		
Configuration	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFNR.		
	The user A and the user C are in network N1 and user C is provided with TIP. The user B is in network N2 and is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=No, "diverting number is released to the diverted-to user"=No, "served user receives notification that the call has been forwarded" =no); Served user communication retention on invocation of diversion (forwarding or deflection)= No.		
Selection criteria:	Call forwarding by the network.		
	Call forwarding not reply supported.		
	CF Notifications supported.		
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is not notified of call diversion and not informed of the diverted-to number (user C has presentation allowed – TIP) and user C is not informed of the forwarding number (user B has presentation not allowed).		
	User B is not notified of call diversion		
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;		
0	Case c) Supported: 100 rel and precondition.		
Comments:	Continue de		
	Continued:		



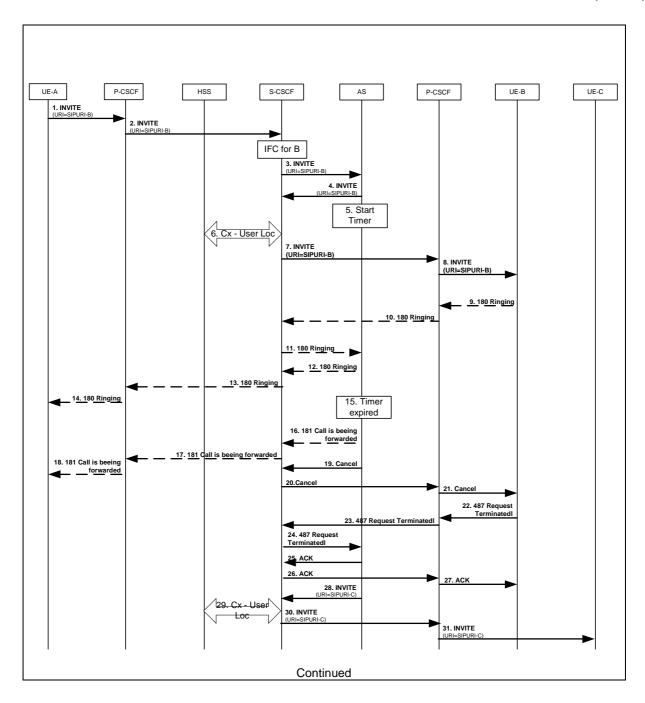


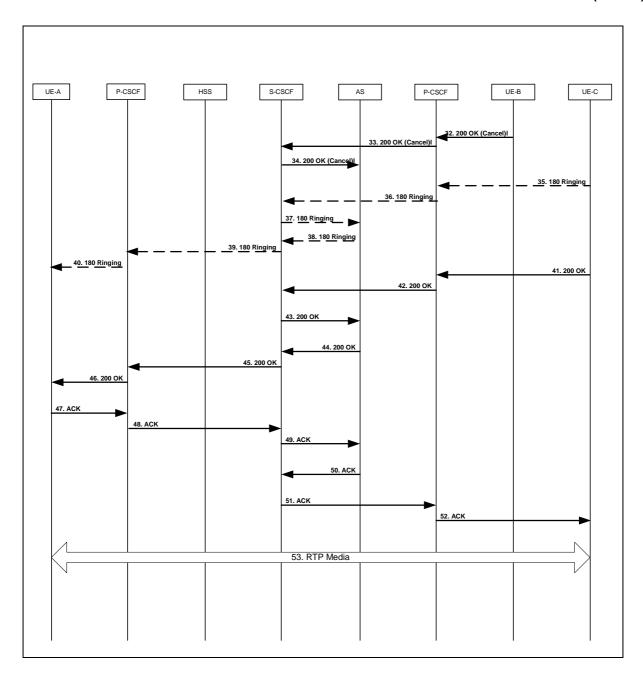
SSS_XXSSCFNR 06	NGN reference to: [17]	
TSS reference:	SIP-SIP/Supplementary_services/CFNR/	
Configuration	The user A and the user C are in network N1. The user B is provided with CFNR.	
	Served user communication retention on invocation of diversion (forwarding or deflection)= Yes.	
Selection criteria:	CFNR supported:	
	 CF Notifications supported. 	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C.	
	The diverting user accepts the communication after sending the INVITE request, the	
	communication path towards the diverted to user shall be released according to the	
	rules and procedures in RFC 3261 [2].	
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:		
	Continued	



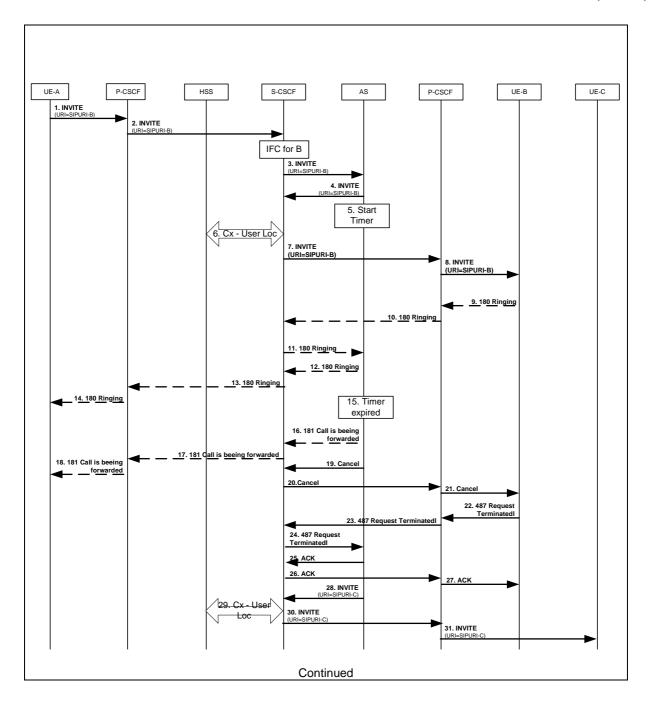


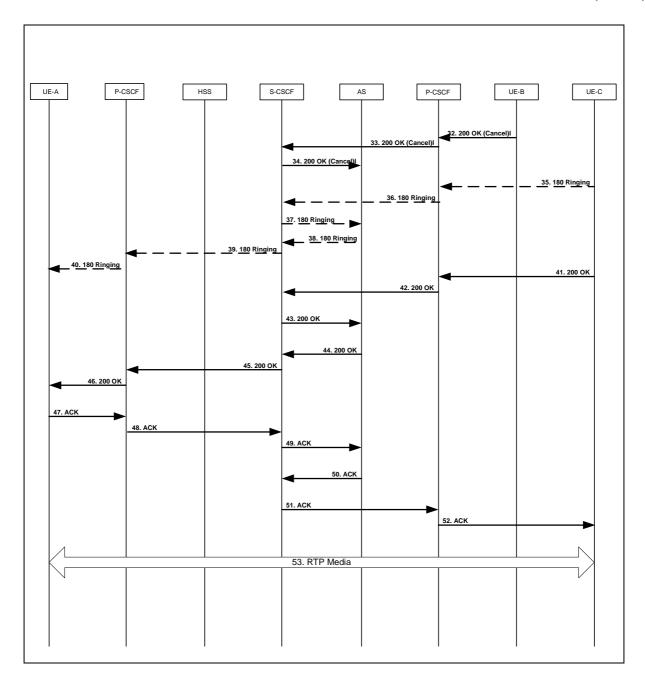
SSS_XXSSCFNR 07	NGN reference to: [17]		
TSS reference:	SIP-SIP/Supplementary_services/CFNR/		
Configuration	The user A and the user C are in network N1. The user B is provided with CFNR		
	("Served user allows the presentation of forwarded to URI to originating user in		
	diversion notification"=Yes, "diverting number is released to the diverted-to user"=Yes,		
	"served user receives notification that the call has been forwarded" =Yes, Served user		
	communication retention on invocation of diversion (forwarding or deflection)= Yes		
Selection criteria:	CFNR supported:		
	CF Notifications supported.		
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C.		
	The diverting user accepts the communication after sending the INVITE request, the		
	communication path towards the diverted to user shall be released according to the		
	rules and procedures in RFC 3261 [2].		
	User A is notified of call diversion and informed of the diverted-to number (user C has		
	presentation allowed - no TIR) and user C is informed of the forwarding number (use		
	B has presentation allowed). User B is notified of call diversion.		
SIP Parameter values:	Dial string parameters options=PIXIT		
en raiameter valuee.	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:			
	Continued		





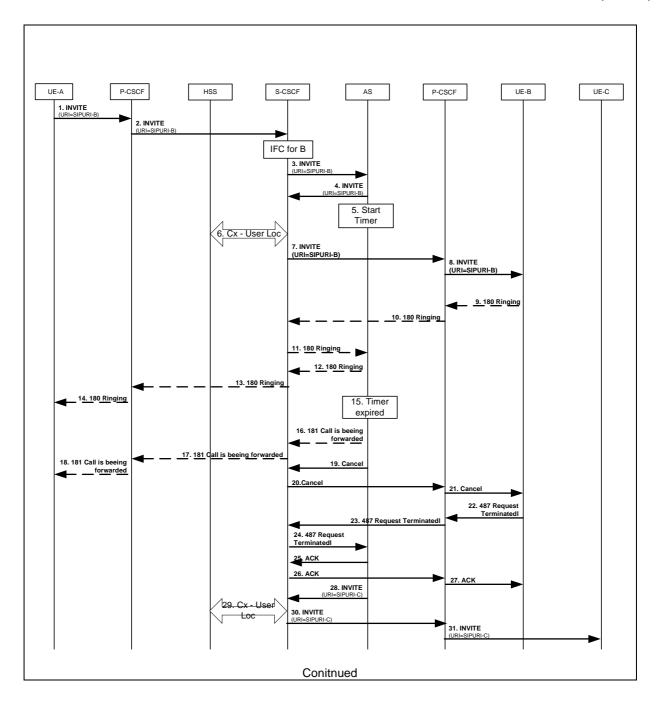
SSS_XXSSCFNR 08	NGN reference to: [17]	
TSS reference:	SIP-SIP/Supplementary_services/CFNR/	
Configuration	The user A and the user C are in network N1. The user B is provided with CFNR	
	("Served user allows the presentation of forwarded to URI to originating user in	
	diversion notification"=Yes, "diverting number is released to the diverted-to user"=Yes,	
	"served user receives notification that the call has been forwarded" =No, Served user communication retention on invocation of diversion (forwarding or deflection)= Yes	
Selection criteria:	CFNR supported:	
	CF Notifications supported.	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C.	
	The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [2].	
	User A is notified of call diversion and informed of the diverted-to number (user C has presentation allowed - no TIR) and user C is informed of the forwarding number (user B has presentation allowed). User B is not notified of call diversion	
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:		
	Continued	

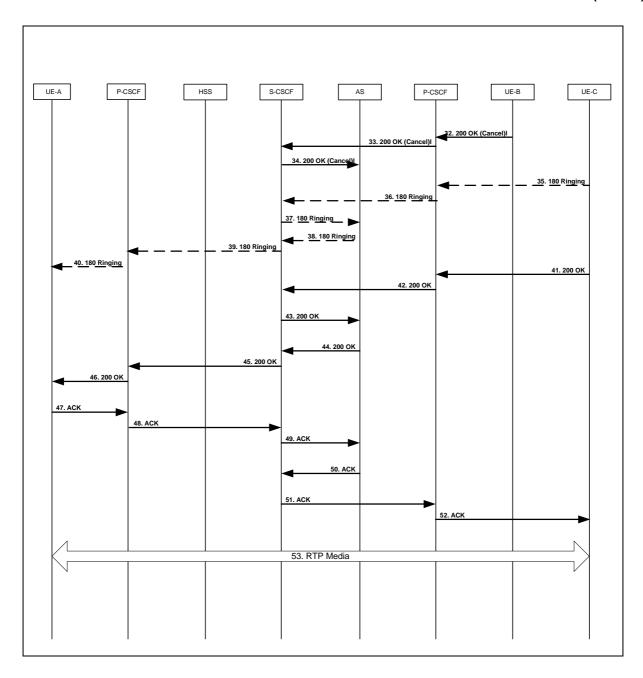




SSS_XXSSCFNR 09	NGN reference to: [17]	
TSS reference:	SIP-SIP/Supplementary_services/CFNR	
Configuration	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting number is released to the diverted-to user"=No, "served user receives notification that the call has been forwarded" =no) Served user communication retention on invocation of diversion (forwarding or deflection)=Yes	
Selection criteria:	 Call forwarding by the network. Call forwarding not reply supported. CF Notifications supported. 	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C. The diverting user accepts the communication after sending the INVITE request, the communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [2]. User A is notified of call diversion and not informed of the diverted-to number (user C has presentation not allowed – TIR) and user C is not informed of the forwarding number (user B has presentation not allowed). User B is not notified of call diversion.	
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:		

SSS_XXSSCFNR 10	NGN reference to: [17]	
TSS reference:	SIP-SIP/Supplementary_services/CFNR	
Configuration	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFNR	
	The user A and the user C are in network N1 and user C is provided with TIP. The user B is in network N2 and is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=No, "diverting number is released to the diverted-to user"=No, "served user receives notification that the call has been forwarded" =no); Served user communication retention on invocation of diversion (forwarding or deflection)= Yes.	
Selection criteria:	Call forwarding by the network.	
	Call forwarding not reply supported.	
	CF Notifications supported.	
Test purpose:	Ensure that when user A calls user B, if unanswered, the call is forwarded to user C	
	The diverting user accepts the communication after sending the INVITE request, the	
	communication path towards the diverted to user shall be released according to the rules and procedures in RFC 3261 [2]. User A is not notified of call diversion and not informed of the diverted to purple (user C has presentation and allowed. TID) and user C is not	
	of the diverted-to number (user C has presentation not allowed – TIR) and user C is not informed of the forwarding number (user B has presentation not allowed).	
	User B is not notified of call diversion.	
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:		
	Continued	





SSS_XXSSCFNR 15	IMS reference to: [17]	NGN reference to: [17]
TSS reference:	SIP-SIP/Supplementary_services/CFNR	
Configuration	The user B is in network N2 and is provided with CFNR, Served user communication retention on invocation of diversion (forwarding or deflection)= No [Clear call to the served user on invocation of call diversion].	
Selection criteria:	 Call forwarding by the network CFNR supported. user C is user determined user 	
Test purpose:	To verify that a call is released correctly if CFNR was not successful . User A calls user B, the call is forwarded to user C who is user determined user busy.	
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel ar	
Comments:		

SSS_XXSSCFNR 16	IMS reference to: [17]	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNR	
Configuration	The user B is in network N2 and is provided with CFNR, Served user communication retention on invocation of diversion (forwarding or deflection)= No	
Selection criteria:	Call forwarding by the networkCFNR supported.user C is network determine	
Test purpose:	To verify that a call is released correctly if CFNR was not successful. User A calls user B, the call is forwarded to user C who is network determined user busy.	
SIP Parameter values:	Dial string parameters options=PIX TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel ar	
Comments:		

SSS_XXSSCFNR 17	NGN reference to: [17]		
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNR		
Configuration	The user B is in network N2 and is provided with CFNR		
	Served user communication retention on invocation of diversion (forwarding or deflection)= Yes		
Selection criteria:	Call forwarding by the network.		
	CFNR supported.		
	user C is user determined user busy.		
Test purpose:	User A calls user B, the call is forwarded to user C who is user determined user busy.		
	The forwarding user User B continues to alert.		
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:			

SSS_XXSSCFNR 18	NGN reference to: [17]	
TSS reference:	SIP-SIP/Supplementary_services/CFNR	
Configuration	The user B is in network N2 and is provided with CFNR, Served user communication retention on invocation of diversion (forwarding or deflection)= Yes	
Selection criteria:	 Call forwarding by the network. CFNR supported. user C is network determined user busy. 	
Test purpose:	User A calls user B, the call is forwarded to user C who is network determined user busy. The forwarding user User B continues to alert.	
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:		

SSS_XXSSCFNR 18	NGN reference to: [17]										
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNR										
Configuration	The user B is in network N2 and is provided with CFNR,										
	retention on invocation of diversion (forwarding or deflec	tion)= Yes									
Selection criteria:	Call forwarding by the network.										
	! •	CFNR supported.									
	user C is network determined user busy. The second s										
Test purpose:	The served user subscribes to the CFNR simulation ser Additionaly an announcement is played.	vice; originating user is notified.									
SIP Parameter values:	When Communication Diversion occurs and if the notific originating user is supported then a 181 (Call Is Being F towards the originating user. Additional the AS initiates an announcement to be included order to inform about the diversion. Announcements shapprocedures as are described in TS 183 028 [18]. 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 SIP header values:	orwarded) response shall be sent ded towards the calling user in									
_	181 Call is Being Forwarded:										
Comments:	CLIT	0.1D.#2									
SIP#1	SUT SIP (served										
INVITE 1	→ INVITE	user)									
	INVITE → INVITE										
180 Ringing	← 180 Ringing ← 180 Ring CANCEL → CANCEL										
181 Call is Being Forwa	rded ← 181 Call is Being Forwarded	_									
	INVITE →	→ INVITE									
	180 Ringing ←	180 Ringing									
180 Ringing	← 180 Ringing 200 OK (INVITE) ←	← 200 OK (INVITE)									
200 OK (INVITE) ACK	← 200 OK (INVITE)	→ ACK									
ACK	Communication	7 ACK									
BYE	→	→ BYE									
200 OK (BYE)	←	← 200 OK (BYE)									

SSS_XXSSCFNR 19	NGN referer	nce to: [17] clause 4.5.2	.6.3			
TSS reference:		lementary_services/CFN				
Configuration		twork N2 and is provided				communication
		ation of diversion (forward	ding or	deflection)= Ye	es	
Selection criteria:		ding by the network.				
	CFNR supp					
	user C is ne	etwork determined user b	usy.			
Test purpose:		ubscribes to the CFNR si rerting user using the ME				
	including the inform	the diverting user is regis nation where the call is d er value that can be provi	iverted	to. The MESS		
SIP Parameter values:	Dial string paramet TYPE_SDP= PIXIT PIXIT for supported Case a) No 1 Case b) Supp	ters options=PIXIT T; d header: 00 rel; ported: 100 rel; ported: 100 rel and preco				
Comments:	3					
SIP#1	'	SUT		SIP#2		SIP#3
INVITE 1	→ INVITE	-		(served user)		
INVITE I	7 INVIII	= INVITE	-	NVITE		
180 Ringing	←	180 Ringing CANCEL INVITE	← 1		_	INVITE
		MESSAGE 200 OK	→ N		7	INVIIE
190 Dinging	← 180 Ri	180 Ringing	←		←	180 Ringing
180 Ringing		200 OK (INVITE)	←		←	200 OK (INVITE)
200 OK (INVITE)		K (INVITE)			_	101
ACK	→	Communication			→	ACK
BYE 200 OK (BYE)	→	Communication			→	BYE 200 OK (BYE)

SSS_XXSSCFNR 21	NGN re	ference to: [17] clause 4.5.2.6	.3							
TSS reference:		SIP-SIP/Supplementary_services/CFNR									
Configuration	The user B is in network N2 and is provided with CFNR, Served user communication										
		retention on invocation of diversion (forwarding or deflection)= Yes									
Selection criteria:	II.	Call forwarding by the network.									
		CFNR supported. user C is network determined user busy.									
Toot nurnoon:		user C is network determined user busy. The served user subscribes to the CFNR simulation service; Indication of communication									
Test purpose:					SAGE request wh						
		ication is reques		0	or to 2 roquoot with	on a	non catgonig				
		,									
							est after the diverting				
		initiated a new	outgoing commu	nica	tion the information	n wh	ere the call is diverted				
SIP Parameter values:	to.	g parameters or	otions_DIVIT								
SIF Farameter values.		g parameters of DP= PIXIT;	ווטווא=רואוו								
		supported head	der:								
	Case	a) No 100 rel	1								
		b) Supported:									
	Case	c) Supported: ler values:	: 100 rel and pre	conc	lition.						
	II.	ier values. is Being Forwar	ded:								
Comments:	101 04	io Boilig Forwar	 								
SIP#1		S	UT		SIP#2		SIP#3				
	_				(served user)						
INVITE 1	→	INVITE	INIVITE	_	INVITE						
180 Ringing	←				180 Ringing						
Too ranging	-				CANCEL						
			INVITE	→		→	INVITE				
			MESSAGE								
			200 OK		200 OK	_	100 Dinging				
180 Ringing	←	180 Ringing	180 Ringing	_		+	180 Ringing				
Too ranging	•		00 OK (INVITE)	←		←	200 OK (INVITE)				
200 OK (INVITE)	←	200 OK (INVIT					, ,				
ACK	→	•				→	ACK				
BYE	→	Co	mmunication			_	BYE				
200 OK (BYE)	-						200 OK (BYE)				
Communication	-					-	200 311 (212)				
		INVITE 3		→		→	INVITE				
		180 Ringing		+	100 B' '	←	180 Ringing				
				+	180 Ringing 200 OK	+	200 OK (INVITE)				
				•	(INVITE)		200 OK (INVITE)				
			MESSAGE	→	MESSAGE						
i			200 OK		200 OK						
			200 010		ACK		ACK				

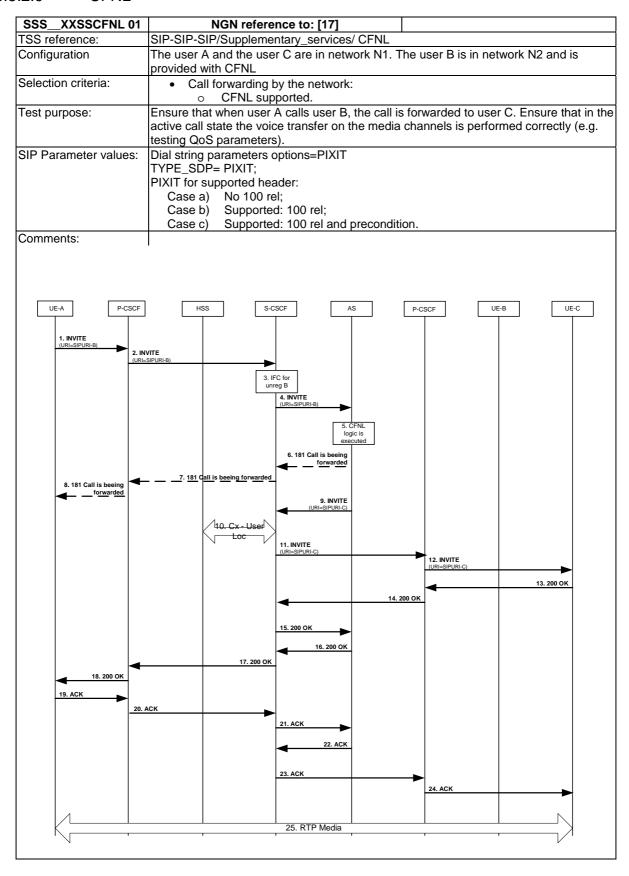
SSS_XXSSCFNR 22	NGN reference to: [17] clause	4.5.2.6.3								
TSS reference:	SIP-SIP-SIP/Supplementary_service	s/CFNR								
Configuration	The user B is in network N2 and is p			communication						
	etention on invocation of diversion (forwarding or deflection)= Yes									
Selection criteria:	Call forwarding by the network.									
	 CFNR supported. 									
	 user C is network determined 									
Test purpose:	The served user subscribes to the C		,	n of communication						
	diversion to the diverting user using	a voicemail or r	nail system.							
				de Maissansil en						
	Ensure that when communication div Message mail system including the in									
SIP Parameter values:	Dial string parameters options=PIXIT		ite title call is diver	ied io.						
SIF Farameter values.	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#1	SUT		SIP#2	SIP#3						
IND (ITT 4	N INDUITE	(se	erved user)							
INVITE 1	→ INVITE	VITE → INV	UTC							
180 Ringing		nging 🗲 180								
180 Kinging			NCEL							
		VITE →	→	INVITE						
	Voicemail or Message		cemail or							
			ssage mail							
		sys	tem							
	180 Rii	nging 🗲	←	180 Ringing						
180 Ringing	← 180 Ringing		_							
000 OK (INI) (ITE)	200 OK (IN)	/ITE) ←	←	200 OK (INVITE)						
200 OK (INVITE) ACK	← 200 OK (INVITE)		→	ACK						
AUN	Communica	ion	7	AUN						
BYE	→		→	BYE						
200 OK (BYE)	É			200 OK (BYE)						
	-		_							
The test case can not be	e tested with an ATS									

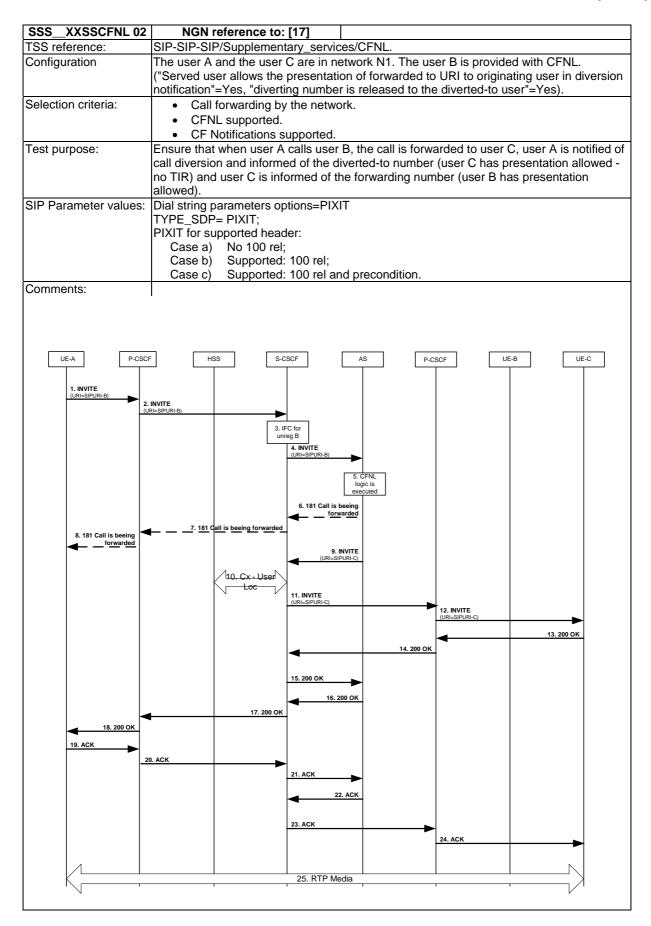
SSS_XXSSCFNR 23	NGN refe	rence to: [17] clause 4.5.2.6.3								
TSS reference:	SIP-SIP-SIF	P/Supplementary_services/CFN	R							
Configuration	The user B i	is in network N2 and is provided	with	n CFNR, Served us	ser communication					
	retention on	etention on invocation of diversion (forwarding or deflection)= Yes								
Selection criteria:	 Call f 	forwarding by the network.								
		R supported.								
		C is network determined user b								
Test purpose:		user subscribes to the CFNR si								
	diversion to	the diverting user using a voice	mail	or mail system red	quested by a timer.					
		when communication diversion								
		ail system request including the								
	Voicemail or by the user.	r Message mail system shall be	sen	t due to a timer val	lue that can be provided					
SIP Parameter values:		arameters options=PIXIT								
	TYPE SDP									
	PIXIT for su	pported header:								
	Case a)	No 100 rel;								
	Case b)									
	Case c)	Supported: 100 rel and preco	nditi	on.						
Comments:										
SIP#1										
311#1		SUT		SIP#2	SIP#3					
	_			SIP#2 (served user)	SIP#3					
INVITE 1	→	INVITE	_	(served user)	SIP#3					
INVITE 1		INVITE INVITE		(served user)	SIP#3					
	→	INVITE INVITE 180 Ringing	←	(served user) INVITE 180 Ringing	SIP#3					
INVITE 1		INVITE INVITE 180 Ringing CANCEL	← →	(served user)						
INVITE 1		INVITE INVITE 180 Ringing	← →	(served user) INVITE 180 Ringing	SIP#3 → INVITE					
INVITE 1		INVITE INVITE 180 Ringing CANCEL INVITE	← →	(served user) INVITE 180 Ringing	→ INVITE					
INVITE 1		INVITE INVITE 180 Ringing CANCEL	← → →	(served user) INVITE 180 Ringing						
INVITE 1 180 Ringing	+	INVITE INVITE 180 Ringing CANCEL INVITE 180 Ringing	← → → ←	(served user) INVITE 180 Ringing	→ INVITE					
INVITE 1 180 Ringing	+	INVITE INVITE 180 Ringing CANCEL INVITE 180 Ringing 180 Ringing 200 OK (INVITE) 200 OK (INVITE)	← → → ←	(served user) INVITE 180 Ringing	→ INVITE ← 180 Ringing					
INVITE 1 180 Ringing 180 Ringing	+	INVITE INVITE 180 Ringing CANCEL INVITE 180 Ringing 180 Ringing 200 OK (INVITE) 200 OK (INVITE) Voicemail or Message mail	←→→ ← ←	(served user) INVITE 180 Ringing CANCEL Voicemail or	→ INVITE ← 180 Ringing					
INVITE 1 180 Ringing 180 Ringing	+	INVITE INVITE 180 Ringing CANCEL INVITE 180 Ringing 180 Ringing 200 OK (INVITE) 200 OK (INVITE)	←→→ ← ←	(served user) INVITE 180 Ringing CANCEL Voicemail or Message mail	→ INVITE ← 180 Ringing					
INVITE 1 180 Ringing 180 Ringing 200 OK (INVITE)	÷ ÷	INVITE INVITE 180 Ringing CANCEL INVITE 180 Ringing 180 Ringing 200 OK (INVITE) 200 OK (INVITE) Voicemail or Message mail	←→→ ← ←	(served user) INVITE 180 Ringing CANCEL Voicemail or	→ INVITE ← 180 Ringing ← 200 OK (INVITE)					
INVITE 1 180 Ringing 180 Ringing	+	INVITE INVITE 180 Ringing CANCEL INVITE 180 Ringing 200 OK (INVITE) 200 OK (INVITE) Voicemail or Message mail system	←→→ ← ←	(served user) INVITE 180 Ringing CANCEL Voicemail or Message mail	→ INVITE ← 180 Ringing					
INVITE 1 180 Ringing 180 Ringing 200 OK (INVITE)	÷ ÷	INVITE INVITE 180 Ringing CANCEL INVITE 180 Ringing 180 Ringing 200 OK (INVITE) 200 OK (INVITE) Voicemail or Message mail	←→→ ← ←	(served user) INVITE 180 Ringing CANCEL Voicemail or Message mail	→ INVITE ← 180 Ringing ← 200 OK (INVITE) → ACK					
INVITE 1 180 Ringing 180 Ringing 200 OK (INVITE) ACK BYE	÷ ÷	INVITE INVITE 180 Ringing CANCEL INVITE 180 Ringing 200 OK (INVITE) 200 OK (INVITE) Voicemail or Message mail system	←→→ ← ←	(served user) INVITE 180 Ringing CANCEL Voicemail or Message mail	 → INVITE ← 180 Ringing ← 200 OK (INVITE) → ACK → BYE 					
INVITE 1 180 Ringing 180 Ringing 200 OK (INVITE)	÷ + + + +	INVITE INVITE 180 Ringing CANCEL INVITE 180 Ringing 200 OK (INVITE) 200 OK (INVITE) Voicemail or Message mail system Communication	←→→ ← ←	(served user) INVITE 180 Ringing CANCEL Voicemail or Message mail	→ INVITE ← 180 Ringing ← 200 OK (INVITE) → ACK					

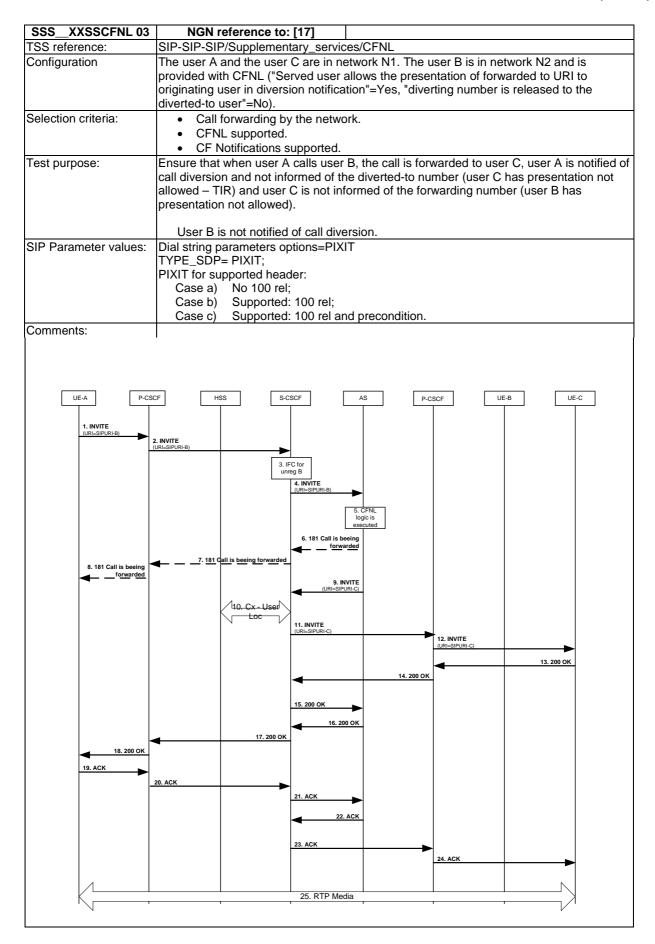
SSS_XXSSCFNR 24	NGN refe	rence to: [17] clause 4.5.2.6.3	3								
TSS reference:		P/Supplementary_services/CFN									
Configuration	The user B	is in network N2 and is provide	d with	n CFNR, Served u	ser o	communication					
		invocation of diversion (forwar	ding	or deflection)= Yes	S						
Selection criteria:		can remaining by the notificial									
		C is network determined user be									
Test purpose:		user subscribes to the CFNR s									
	communica	tion diversion to the diverting us	ser us	sing a voicemail or	r mai	l system.					
		when communication diversion									
		ail system request including the									
		er will be informed periodically where the call is diverted to.	with	roicemail of iviess	age	man system with the					
SIP Parameter values:		parameters options=PIXIT									
SIF Farailleler values.	TYPE_SDP										
		pported header:									
	Case a)										
	Case b)										
	Case c)	Supported: 100 rel and preco	nditi	on							
Comments:	,										
SIP#1		SUT		SIP#2		SIP#3					
				(served user)							
INVITE 1	→	INVITE									
				INVITE							
180 Ringing	←			180 Ringing							
				CANCEL		IN 11 / T.					
		INVITE		V-!!!	→	INVITE					
		Voicemail or Message mail									
		system		Message mail system							
		180 Ringing	←	System	←	180 Ringing					
180 Ringing	←	180 Ringing	•		`	100 Kinging					
100 Kinging	•	200 OK (INVITE)	←		←	200 OK (INVITE)					
200 OK (INVITE)	←	200 OK (INVITE)	_		•	200 011 (1111112)					
ACK	→	200 011 (2)			→	ACK					
		Voicemail or Message mail		Voicemail or							
		system		Message mail system							
		Communication		-							
BYE	→				→	BYE					
200 OK (BYE)	←				←	200 OK (BYE)					
The test case can not	be tested wi	th an ATS									

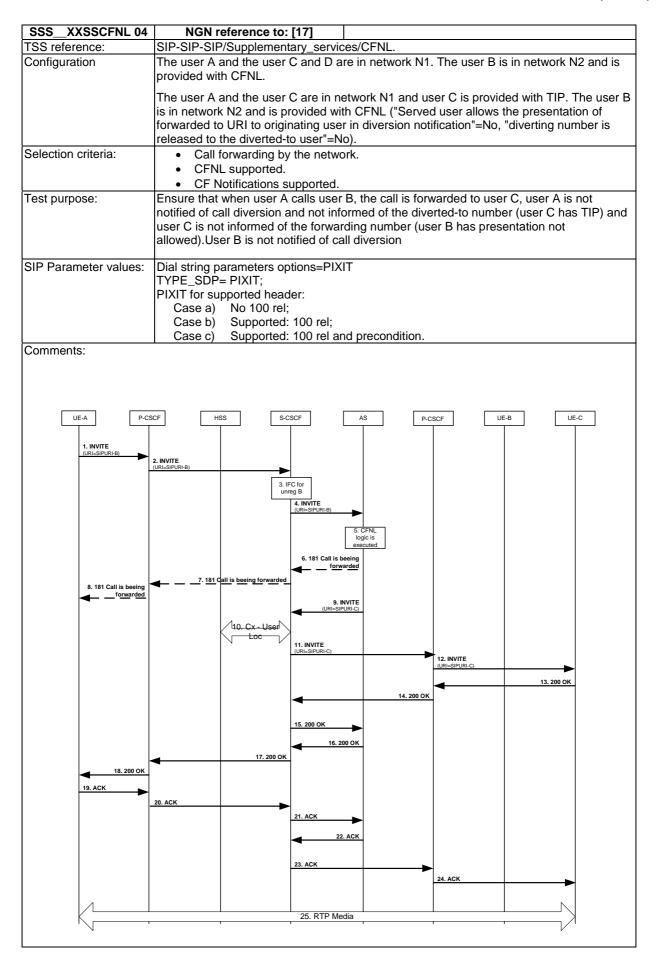
SSS_XXSSCFNR 25	NGN re	ference to: [1	7] clause 4.5.2.6	.3							
TSS reference:			ry_services/CFNI								
Configuration	The user B	is in network N	I2 and is provided	l with	n CFNR, Served us	ser c	communication				
		etention on invocation of diversion (forwarding or deflection)= Yes									
Selection criteria:		Call forwarding by the network.									
	• CFN	CFNR supported.									
	user	C is network of	letermined user b	usy.							
Test purpose:					tion service; Indica						
	communica	tion diversion t	o the diverting us	er us	sing a voicemail or	maı	i system.				
	Encure that	a diverting us	or will be informed	1 va/i+l	h Message Voicen	aail d	or Mossago mail				
					w outgoing comm						
		where the call		a nc	w oatgoing comm	uiiio	auon inc				
SIP Parameter values:		parameters o									
on raramotor values.	TYPE_SDP		p								
		pported heade	er:								
	Case a)										
	Case b)										
	Case c)	Supported:	100 rel and preco	nditio	on						
Comments:											
SIP#1		•	SUT		SIP#2		SIP#3				
INDUITE 4	_	IND ATE			(served user)						
INVITE 1	→	INVITE	INIVITE	_	INIV/ITE						
180 Ringing	←		INVITE		INVITE 180 Ringing						
100 Kinging	•		CANCEL		CANCEL						
			INVITE		ONIVOLL	→	INVITE				
		Voicemail c	or Message mail		Voicemail or	-					
			system		Message mail						
			-		system						
			180 Ringing	←		←	180 Ringing				
180 Ringing	←	180 Ringing	-	_		_					
		2	200 OK (INVITE)	←		←	_00 0.1				
200 OK (INIVITE)	_	200 OK (INI)/I	TC\				(INVITE)				
200 OK (INVITE) ACK	-	200 OK (INVI	I =)			→	ACK				
ACK	7	Co	mmunication			7	ACK				
		INVITE	minumeation	→		→	INVITE				
			r Message mail	→	Voicemail or	-					
			system		Message mail						
			•		system						
							180 Ringing				
						←	200 OK				
					(INVITE)	_	(INVITE)				
The test sees ser = -1.		th on ATC			ACK	→	ACK				
The test case can not l	be tested Wi	ui an ATS									

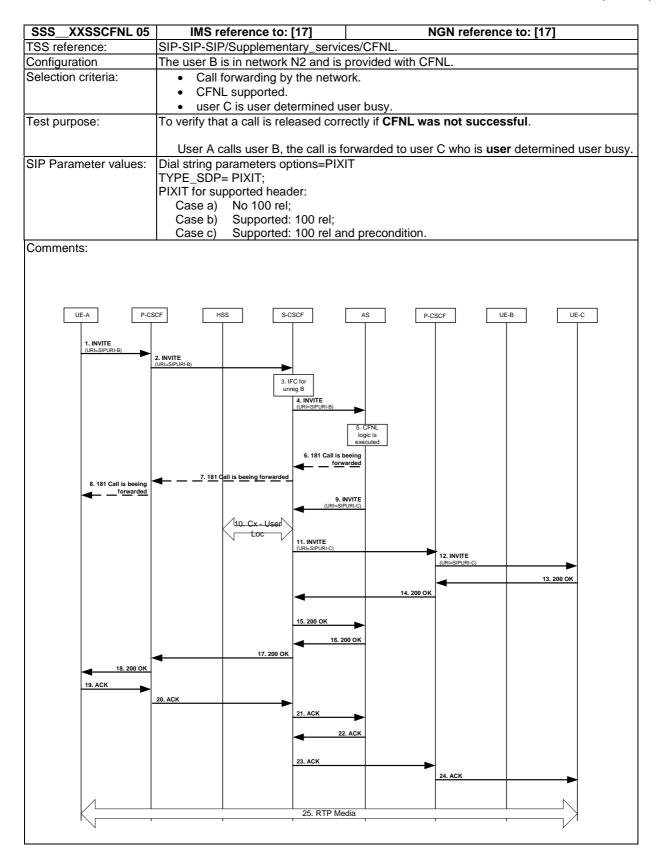
6.5.2.9 CFNL







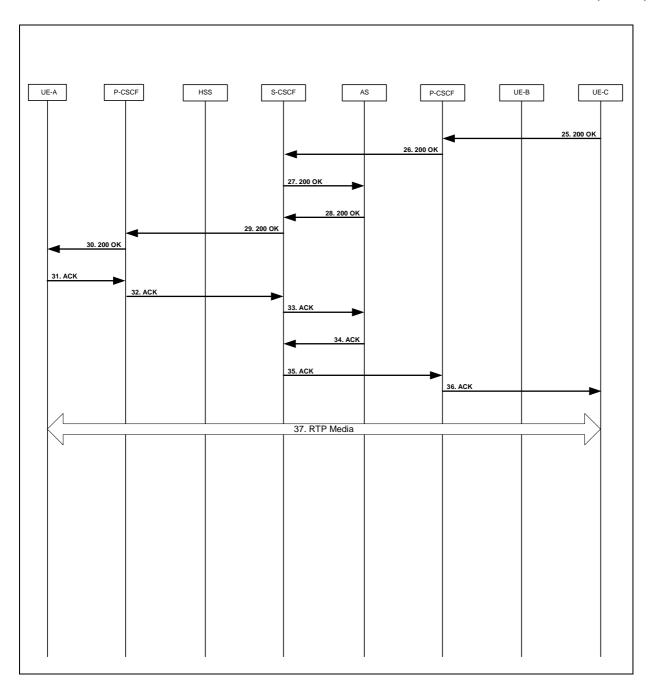


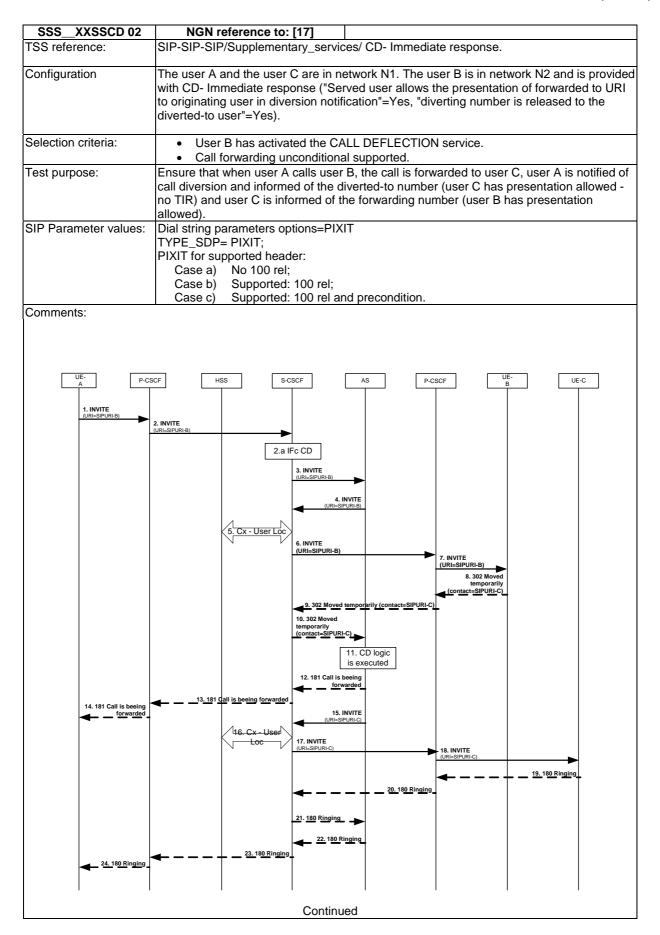


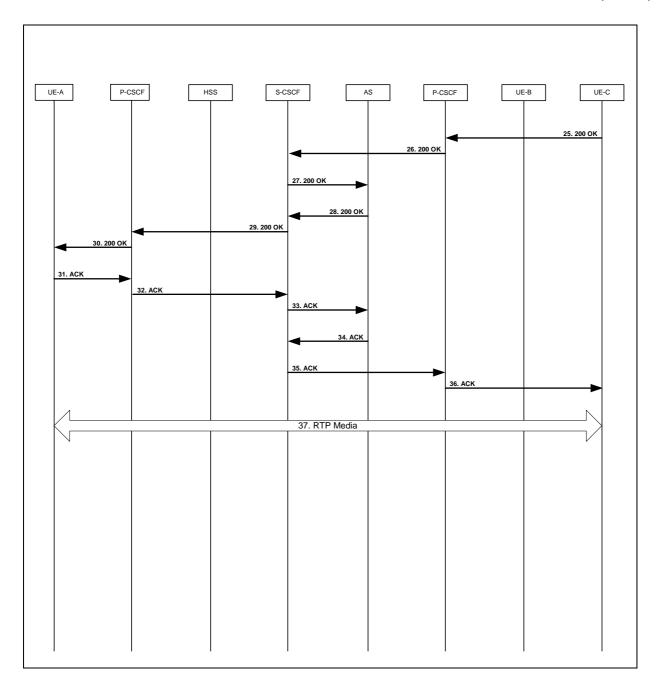
SSS_XXSSCFNL 06	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNL.
Configuration	The user B is in network N2 and is provided with CFNL.
Selection criteria:	Call forwarding by the network.
	CFNL supported.
	user C is network determined user busy.
Test purpose:	To verify that a call is released correctly if CFNL was not successful.
	User A calls user B, the call is forwarded to user C who is network determined user busy.
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	

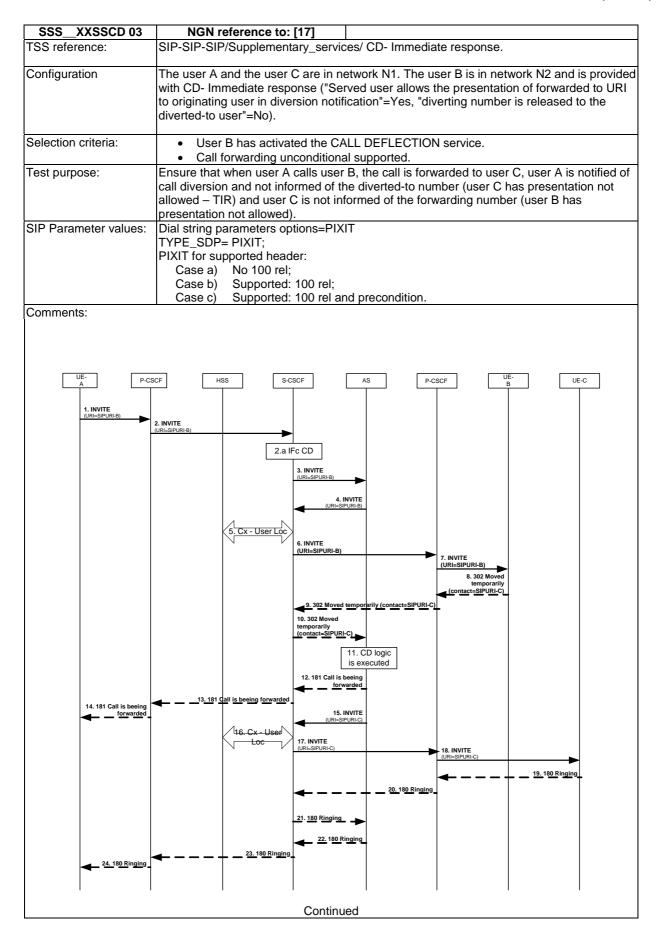
6.5.2.10 CALL DEFLECTION

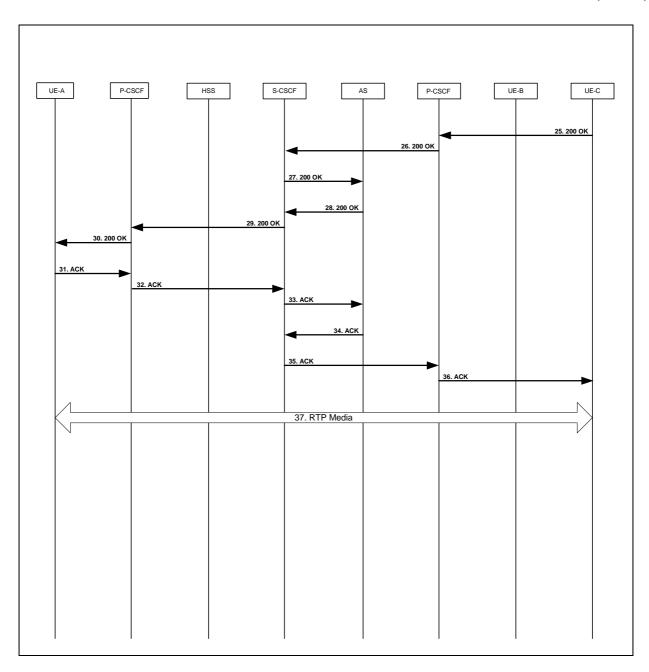
SSS_XXSSCD 01	NGN reference to: [17]								
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CD- Immediate response.								
Configuration	he user A and the user C are in network N1. The user B is in network N2 and is provided ith CD- Immediate response.								
Calaatian anitania.									
Selection criteria:	User B has activated the CALL DEFLECTION service. Call for warding a vector divisional average and described.								
Toot numace	Call forwarding unconditional supported. Ensure that when user A calls user B, the call is forwarded to user C.								
Test purpose: SIP Parameter values:	Dial string parameters options=PIXIT								
oir raidilletei values.	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:									
UE- A P-CS	CF HSS S-CSCF AS P-CSCF UE-C								
1. INVITE (URI=SIPURI-B)									
	2. INVITE (URI=SIPURI-B)								
	2.a IFc CD								
	3. INVITE (URI=SIPURIB)								
	4. INVITE								
	(URI=SIPUR+B)								
	5. Cx - User Loc 6. INVITE								
	(URI=SIPURI-B) 7. INVITE								
	(URI=SIPURI-B) 8. 302 Moved								
	temporarily (contact=SIPURI-C)								
	9. 302 Moved temporarily (contact=SIPURI-C)								
	10. 302 Moved temporarily								
	(contact=SIPURI-C)								
	11. CD logic								
	is executed								
	12. 181 Call is beeing forwarded								
14 181 Call is beeing	13. 181 Qall is beeing forwarded								
forwarded	15. INVITE (URI=SIPURI-C)								
	16. Cx - User								
	LOC 17. INVITE (URL=SIPUR+C) 18. INVITE (URL=SIPUR+C)								
	20, 400 Binding								
	20.180 Ringing								
	21. 180 Ringing								
	22. 180 Ringing								
24. 180 Ringing	◄ — — — — 23.180 Ringing								
1									
	Continued								



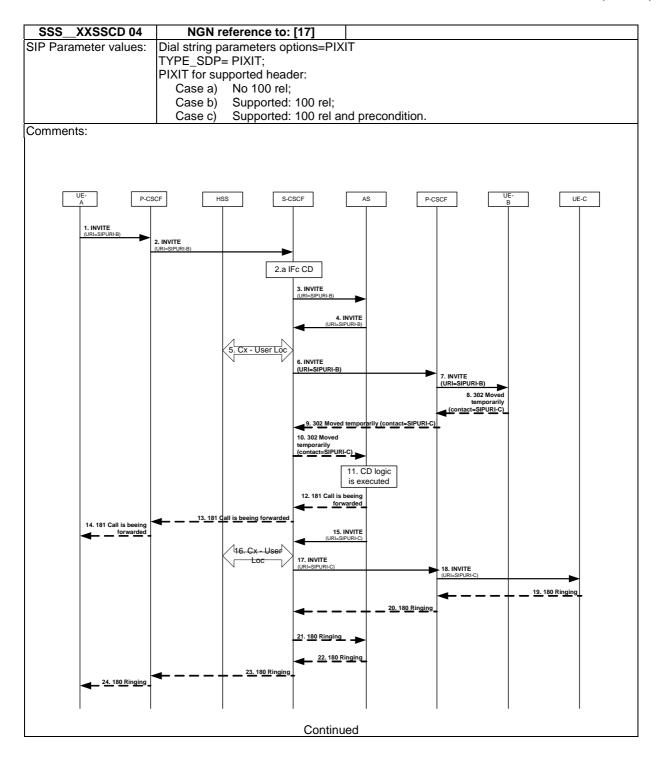


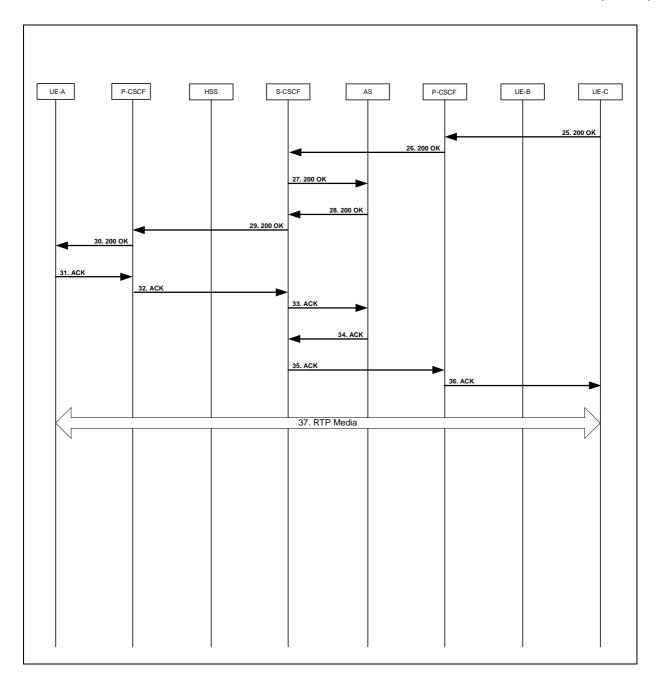






SSS_XXSSCD 04	NGN reference to: [17]	
TSS reference:	SIP-SIP/Supplementary_services/ CD- Immediate response.	
Configuration	The user A and the user C and D are in network N1. The user B is in network N provided with CD- Immediate response.	N2 and is
	The user A and the user C are in network N1 and user C is provided with TIP. is in network N2 and is provided with CFU ("Served user allows the presentation forwarded to URI to originating user in diversion notification"=No, "diverting nurreleased to the diverted-to user"=No).	on of
Selection criteria:	 User B has activated the CALL DEFLECTION service. Call forwarding unconditional supported. 	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C, user A is notified of call diversion and not informed of the diverted-to number (user C has user C is not informed of the forwarding number (user B has presentation not a	s TIP) and





SSS_XXSSCD 05		NGN	reference	to:	[17]								
TSS reference:	SIP-S	IP-SI	P/Supplen	nenta	ary_service	es/ C	D- Immedia	ate r	esponse				
Configuration							ded with CE				ponse		
Selection criteria:	•	Call	forwardin	g und	contitional	supp	orted.						
	•	user C is network determined user busy.											
	•	Use	r B has ac	tivat	ed the CAI	LL D	EFLECTIO	N se	ervice.				
Test purpose:	To ve	rify th	at a call is	rele	ased corre	ectly	f CALL DE	FLE	CTION w	/as	not succe	essful.	
		į	Jser A cal	ls us	er B, the c	all is	forwarded	to u	ser C wh	o is	user det	ermin	ed
			user busy.										
SIP Parameter values:				s opt	ions=PIXI	T							
			P= PIXIT;										
			upported h		er:								
		ase a)			100 1								
		ase b) ase c)			100 rei; 100 rel and	d nre	condition						
Comments:	Uč	136 0)	Suppoi	ieu.	100 lei aili	u pie	condition.						
Comments.	UE 1		I-CSCF		S-CSCF		AS		P-CSCF		UE-B	U	E-C
	INVITE	→	INVITE	→									
	IINVIIE	7	INVIIE	7									
			100 Trying	←		_							
					INVITE 100 Trying	→							
					INVITE	È							
					100 Trying	→	INDUITE						
							INVITE 100 Trying	→					
									INVITE	→			
									100 Trying 302	+			
							302	←	302	•			
					302	→							
					ACK	+	I ACK	→	ACK	→			
					181	(-	71011	-			
	181	←	181	+									
	101	_			INVITE	(
					100 Trying	→						_	
					INVITE	→	486	+			INVITE 486	→	
					486	→	400	•			ACK	→	
	406	_	486	←	486	←	I						
	486 ACK	← →											
			ACK	→									
					ACK ACK	→							
					AOIX	~	ACK	→					
			<u> </u>		<u> </u>								

1	11011	reference	to: [17]							
TSS reference:	SIP-SIP-SII	P/Supplem	nenta	ry_service	s/ Cl	D- Immedia	ite r	esponse	CD	- Immedia	ate
	response.										
	The user B						- Im	nmediate	res	ponse.	
Selection criteria:											
T						FLECTION					
Test purpose:	To verify th	at a call is	reiea	ased corre	Ctiy ii	CALL DE	-LE	CHONV	vas	not succe	SSIUI.
	User A	calls user	B th	e call is fo	rwar	ded to user	Cv	who is ne	twc	rk detern	nined user
	busy.	cano acor	٠,	10 0411 10 10	····	404 10 4001	٠.			in dotom	
SIP Parameter values: [Dial string p	parameters	s opt	ions=PIXI							
	TYPE_SDF	P= PIXIT;									
F	PIXIT for su			er:							
	Case a)			100 1-							
	Case b) Case c)			100 rei; 100 rel and	d nro	condition					
Comments:	Case c)	Guppon	.cu.	TOO TEL ALIC	ı pıcı	ooriuitioi1.					
UE 1	P-C	SCF	S-C	SCF	AS		P-C	SCF	UE	-В	UE-C
IN'	VITE →	INVITE	→								
		100 Trying	+	INVITE	→						
				100 Trying	←						
				INVITE 100 Trying	←						
				100 Trying	- 1	INVITE	→				
						100 Trying	+	INVITE	→		
								100 Trying			
						302	+	302	+		
				302	→	302	•				
				ACK	←	ACK	→	ACK	→		
				181	←	AOR	7	AOR	1		
	181 ←	181	←								
	101			INVITE	+						
				100 Trying INVITE	→						
				INVITE	→	486	+				
		400	,	486	→						
	486 ←	486	+	486	+						
A	ACK →	A CIV	→								
		ACK	7	ACK	→						
				ACK	+	A CIV					
						ACK	→				

6.5.2.11 CONF

6.5.2.11.1 Conference creation

SSS_XXSSCONF_ CRE_001	NGN referen	ce to: [20] clause 5.3.1.3.3					
TSS reference:	SIP-SIP-SIP/Supp	lementary_servicesCONF					
Configuration	CONF						
Selection criteria:	CONF						
Test purpose:	Conference creation by Three-way session creation. REFER request to the user, Conference						
rest purpose.	event package is subscribed.						
	these active sessi		P sessions and wants to join together two of on. The conference event package is not the following steps:				
	Create a co conference	onference at the conference focus factory URI for the three-way ses	by sending an INVITE request with the sion towards the conference focus. are requested to be joined to the three-way				
	session by REFER red	performing the procedures for inv quest to the user.	iting a user to a conference by sending a				
	focus is establishe	ed by SIP#2 and SIP#3.	E was informed concerned the session to the				
SIP Parameter	Dial string parame	ters options=PIXIT					
values:	TYPE_SDP= PIXI	T;					
	PIXIT for supporte						
		100 rel;					
		ported: 100 rel;					
		ported: 100 rel and precondition.					
	INVITE:	Request URI contained the confe					
	200 OK:	"isfocus" feature parameter indic					
		conference URI contained in the					
	SUBSCRIBE:	Request URI contained the confe	erence URI				
		header contains "conference"					
	REFER:	Refer-to header contains the con					
	NOTIFY 1	Event contains conference; Sub					
		message/sipfrag contains SIP/2.					
	NOTIFY 2	Event contains conference; Sub					
		message/sipfrag contains SIP/2.					
	NOTIFY 3	Event contains conference; Sub	•				
		application/conference-info+xml					
	NOTIFY 4	Event contains conference; Sub					
		message/sipfrag contains SIP/2.					
	NOTIFY 5	Event contains conference; Sub					
		message/sipfrag contains SIP/2.					
	NOTIFY 6	Event contains conference; Sub					
		application/conference-info+xml	contains connected, dialled-in				
		Continued					

Comments: SIP#1		Focus			SIP#2		SIP#3
Establishment of session #1		Focus			JIF#Z		JIF#3
INVITE	→			→	INVITE		
180 Ringing	←			-	180 Ringing		
200 OK (INVITE)	←			(200 OK (INVITE)		
ACK	÷			→	ACK		
7.0.1	-			•	71011		
INVITE (sendonly)	→			→	INVITE (sendonly)		
200 OK (recvonly)	+			←	200 OK (recvonly)		
ACK	→			→	ACK		
	Esta	ablishment of sess	ion #2	_	7.0		
INVITE	→					→	INVITE
180 Ringing	←					-	
200 OK (INVITE)	÷					÷	200 OK (INVITE)
ACK (IIVITE)	À					À	ACK
Conference creation	•					•	, tolk
INVITE	→	INVITE					
	•	II V I I L					
200 OK	←	200 OK					
ACK	÷	ACK					
SUBSCRIBE	÷	SUBSCRIBE					
200 OK	ŕ	200 OK					
NOTIFY	È	NOTIFY					
200 OK NOTIFY	→	200 OK NOTIFY	,				
Inviting SIP#2 to the conference		200 010 100 111 1					
REFER	÷nce →			→	REFER		
202 Accepted	-			→	202 Accepted		
ZUZ MUCEPIEU	~	ı	NVITE	/	INVITE		
NOTIEV	←		INVIIE				
NOTIFY 200 OK NOTIFY	→			←	NOTIFY 1 200 OK NOTIFY		
ZUU UK NUTIFT	7	_	200 OK	_	200 OK NOTIFY 200 OK		
		2	ACK		ACK		
NOTIFY	←		ACK	-	NOTIFY 2		
200 OK NOTIFY	→			→	200 OK NOTIFY		
BYE	→ →			→ →	BYE		
	→			7 ←			
200 OK (BYE)	~			~	200 OK (BYE)		
NOTIFY	←	NOTIFY 3					
200 OK NOTIFY	À	200 OK NOTIFY	,				
200 01011111		ting SIP#3 to the c		CE			
REFER	→		,JIII GI GI	50		→	REFER
202 Accepted	-						202 Accepted
202 Accepted	*	ı	NVITE	~			INVITE
NOTIFY	←	'		•		-	
200 OK NOTIFY	À					÷	
200 OK NOTH 1	•	2	200 OK	→		→	
		2	ACK			-	ACK
NOTIFY	←		, .	-		÷	
200 OK NOTIFY	À					÷	
BYE	÷					÷	
200 OK (BYE)	-					-	200 OK (BYE)
ZOO OR (BTE)	~					~	ZOU ON (DIE)
NOTIFY	←	NOTIFY 6					
	•						
200 OK	→	200 OK					

SSS_XXSSCONF_C	NGN reference to:
RE_002	[20] clause 5.3.1.3.3
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.
Configuration	CONF
Selection criteria:	CONF
Test purpose:	Conference creation by Three-way session creation. REFER request to the user, Conference event package not subscribed.
	The conference participant is participating in two SIP sessions and wants to join together two of these active sessions to a so-called three-way session. The conference event package is not subscribed. The conference participant shall perform the following steps: Create a conference at the conference focus by sending an INVITE request with the conference factory URI for the three-way session towards the conference focus. Perform for each of the active sessions, that are requested to be joined to the three-way session by performing the procedures for inviting a user to a conference by sending a REFER request to the user.
	The SIP#1 releases the session 1 and 2 after the UE was informed concerned the session to the focus is established by SIP#2 and SIP#3.
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. SIP header values: INVITE: Request URI contained the conference factory URI. 200 OK: "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. REFER: Refer-to header contains the conference URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. NOTIFY 3 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 4 Event contains conference; Subscription-State contains active
	message/sipfrag contains SIP/2.0 200 OK.
	Continued

Comments: SIP#1	Focus	SIP#2	SIP#3
· · · ·	Focus	31P#2	5IP#3
Establishment of session #1	_		
INVITE	→		
180 Ringing	←	3 3	
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
1	7		7 ACK
Conference creation			
INVITE	→ INVITE		
200 OK	← 200 OK		
ACK	→ ACK		
Inviting SIP#2 to the confere	- 7.0.1		
REFER	→ -	REFER	
	É		
202 Accepted	=		
NOTIFY (INVITE •		
NOTIFY	(
200 OK NOTIFY	→		
	200 OK 🗦		
	ACK €	- ACK	
NOTIFY	(NOTIFY 2 	
200 OK NOTIFY	→ -	200 OK NOTIFY	
BYE	→	BYE	
200 OK (BYE)	←	200 OK (BYE)	
	Inviting SIP#3 to the conferen	ce	
REFER	→		→ REFER
202 Accepted	←		€ 202 Accepted
2027.000ptod	INVITE •	_	← INVITE
NOTIFY	+		← NOTIFY 3
200 OK NOTIFY	`		→ 200 OK NOTIFY
200 OK NOTH 1	200 OK =		→ 200 OK NOTIFI
NOTIEV	ACK	=	← ACK
NOTIFY	(← NOTIFY 4
200 OK NOTIFY	→		→ 200 OK NOTIFY
BYE	→		→ BYE
200 OK (BYE)	←		← 200 OK (BYE)

SSS_XXSSCONF_C	NGN reference to:
RE_003	[20] clause 5.3.1.3.3
TSS reference:	SIP-SIP/Supplementary_services/ CONF.
Configuration	CONF
Selection criteria:	CONF
Test purpose:	Conference creation by Three-way session creation. REFER request to the focus, Conference event package subscribed.
	The conference participant is participating in two SIP sessions and wants to join together two of these active sessions to a so-called three-way session. The conference event package is subscribed. The conference participant shall perform the following steps: • Create a conference at the conference focus by sending an INVITE request with the conference factory URI for the three-way session towards the conference focus. • Perform for each of the active sessions, that are requested to be joined to the three-way session by performing the procedures for inviting a user to a conference by sending a REFER request to the focus. The SIP#1 releases the session 1 and 2 after the UE was informed concerned the session
	to the focus is established by SIP#2 and SIP#3.
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. INVITE: Request URI contained the conference factory URI. 200 OK: "isfocus" feature parameter indicated in Contact header field conference URI contained the conference URI. SUBSCRIBE: Request URI contained the conference URI. REFER 1: Refer-to header contains the URI of user#2. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. NOTIFY 3 Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-out. REFER 2: Refer-to header contains the URI of user#3. NOTIFY 4 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 5 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. NOTIFY 6 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK.
	application/conference-info+xml contains connected, dialled-out.
	Continued

Comments: SIP#1	Focus		SIP#2	SIP#3
Establishment of session #1 INVITE 180 Ringing 200 OK (INVITE) ACK		→	INVITE 180 Ringing 200 OK (INVITE) ACK	311 #3
INVITE (sendonly) 200 OK (recvonly) ACK INVITE 180 Ringing 200 OK (INVITE) ACK Conference creation		→ ← → #2	INVITE (sendonly) 200 OK (recvonly) ACK	→ INVITE ← 180 Ringing ← 200 OK (INVITE) → ACK
INVITE 200 OK ACK SUBSCRIBE 200 OK Inviting SIP#2 to the conference REFER 1 202 Accepted NOTIFY 200 OK NOTIFY NOTIFY 200 OK NOTIFY BYE	→ REFER ← 202 Accepted NOTIFY 1 → 200 OK NOTIFY	INVITE → 200 OK ← ACK →	INVITE 200 OK ACK BYE	
200 OK (BYE) NOTIFY 200 OK NOTIFY REFER 2 202 Accepted NOTIFY 200 OK NOTIFY NOTIFY 200 OK NOTIFY BYE 200 OK (BYE)	 NOTIFY 3 → 200 OK NOTIFY Inviting SIP#3 to the conf → REFER ← 202 Accepted ⊢ NOTIFY 4 → 200 OK NOTIFY 	←	200 OK (BYE)	 → INVITE ← 200 OK → ACK → BYE ← 200 OK (BYE)
NOTIFY 200 OK	NOTIFY 6→ 200 OK			. ,

SSS_XXSSCONF_C	NGN reference to:
RE_004	[20] clause 5.3.1.3.3
TSS reference:	SIP-SIP/Supplementary_services/ CONF.
Configuration	CONF
Selection criteria:	CONF
Test purpose:	Conference creation by Three-way session creation. REFER request to the focus, Conference event package not subscribed.
	 The conference participant is participating in two SIP sessions and wants to join together two of these active sessions to a so-called three-way session. The conference event package is not subscribed. The conference participant shall perform the following steps: Create a conference at the conference focus by sending an INVITE request with the conference factory URI for the three-way session towards the conference focus. Perform for each of the active sessions, that are requested to be joined to the three-way session by performing the procedures for inviting a user to a conference by sending a REFER request to the focus. The SIP#1 releases the session 1 and 2 after the UE was informed concerned the session to the focus is established by SIP#2 and SIP#3.
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.
	INVITE: Request URI contained the conference factory URI.
	200 OK: "isfocus" feature parameter indicated in Contact header field
	conference URI contained in the Contact header field.
	REFER 1: Refer-to header contains the URI of user#2.
	NOTIFY 1 Event contains conference ; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying.
	NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK.
	REFER 2: Refer-to header contains the URI of user#3.
	NOTIFY 3 Event contains conference ; Subscription-State contains active
	message/sipfrag contains SIP/2.0 100 Trying.
	NOTIFY 4 Event contains conference ; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK.

→ ← ← →	Focus		→	SIP#2 INVITE		SIP#3
→ ← ← →			_			
← ← →			_			
← →			←			
→				180 Ringing		
_			←	200 OK (INVITE)		
_			→	ACK		
7			→	INVITE (sendonly)		
←			←	200 OK (recvonly)		
→			→			
Estal	olishment of session	n #2		7.0		
					→	INVITE
						180 Ringing
						200 OK (INVITE)
→					→	ACK
→	INVITE					
←	200 OK					
→						
_	71011					
	DEEED					
	202 Accepted	INI\/ITE	_	INI\/ITE		
_	NOTICVA	IINVIIE	7	IINVIIE		
7	200 OK NOTIFY		_	222 214		
_		ACK	→	ACK		
	200 OK NOTIFY					
_			-			
←			←	200 OK (BYE)		
Inviti		nference				
→	REFER					
←	202 Accepted					
	•	INVITE	→		→	INVITE
←	NOTIFY 3					
→	200 OK NOTIFY					
			←		←	200 OK
					→	ACK
←	NOTIFY 4		-		-	
	-					
	200 OK NOTH I				~	BYE
-					-	200 OK (BYE)
_	→ Estal → Estal → Conce →	Establishment of session Help to the content of session Help	Establishment of session #2 Here	Establishment of session #2 Head of the session #2 H	→ ACK Establishment of session #2 → HNVITE ← 200 OK → ACK Ence → REFER ← 202 Accepted	→ ACK Establishment of session #2 → ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ← ←

SSS_XXSSCONF_C	NGN reference to:					
RE_005 TSS reference:	[20] clause 5.3.1.3.3 SIP-SIP-SIP/Supplementary services/ CONF.					
Configuration	CONF					
Selection criteria:	CONF					
Test purpose:	Conference creation by Three-way session creation. REFER request to the user, Replaces method is used.					
	The conference participant is participating in two SIP sessions and wants to join together two of these active sessions to a so-called three-way session. The replaces method is used to terminate previous individual sessions. The conference participants shall perform the following steps:					
	 Create a conference at the conference focus by sending an INVITE request with the conference factory URI for the three-way session towards the conference focus. Perform for each of the active sessions, that are requested to be joined to the three- 					
	way session by performing the procedures for inviting a user to a conference by sending a REFER request to the user.					
	 The SIP#2 and SIP#3 terminates the session 1 and 2 as indicated in the Replaces header received in the INVTE from the focus. 					
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					
	SIP header values:					
	INVITE: Request URI contained the conference factory URI.					
	200 OK: "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field.					
	REFER1: Refer-to header contains the conference URI and Replaces header for session 1.					
	Refer-To: <sip:conference method="INVITE" uri?replaces="call-id1%3Bto-tagsession1%3Bfrom-tagSession1;">.</sip:conference>					
	BYE 1: Call-ID: call-id1/ To:; tag=session1/ From:;tag=Session1. REFER2: Refer-to header contains the conference URI and Replaces header for					
	session 2.					
	Refer-To: <sip:conference uri?replaces="call-id2%3Bto-</td"></sip:conference>					
	tagsession2%3Bfrom-tagSession2; method=INVITE>.					
	BYE 2: Call-ID: call-id2/ To:; tag=session2/ From:;tag=Session2.					

Comments:	_		
SIP#1	Focus	SIP#2	SIP#3
Establishment of session #			
INVITE	→	→ INVITE	
180 Ringing	←	← 180 Ringing	
200 OK (INVITE)	←	← 200 OK (INVITE)	
ACK '	→	→ ACK	
INVITE (sendonly)	→	→ INVITE (sendonly)	
200 OK (recvonly)	É	€ 200 OK (recvonly)	
ACK	÷	→ ACK	
ACK	Establishment of session #2	- ACK	
INIV/ITE	Stabilistifferit of Session #2		→ INI\/ITE
INVITE			→ INVITE
180 Ringing	(← 180 Ringing
200 OK (INVITE)	(← 200 OK (INVITE)
ACK	→		→ ACK
Conference creation			
INVITE	→ INVITE		
200 OK	← 200 OK		
ACK	→ ACK		
Inviting SIP#2 to the confe			
REFER 1	→	→ REFER	
202 Accepted	É	€ 202 Accepted	
202 Accepted	_	← INVITE	
NOTIEV			
NOTIFY	(← NOTIFY	
200 OK NOTIFY	→	→ 200 OK NOTIFY	
		→ 200 OK	
		← ACK	
NOTIFY	←	← NOTIFY	
200 OK NOTIFY	→	→ 200 OK NOTIFY	
BYE	←	← BYE 1	
200 OK (BYE)	→	→ 200 OK (BYE)	
, ,		,	
	Inviting SIP#3 to the conference		
REFER 2	→		→ REFER
202 Accepted	←		€ 202 Accepted
	INVITE	←	← INVITE
NOTIFY	+	•	← NOTIFY
200 OK NOTIFY	-		→ 200 OK NOTIFY
ZOO OK NOTIFT	=	_	
	200 OK		→ 200 OK
NOTIFY	ACK	~	← ACK
NOTIFY	(← NOTIFY
200 OK NOTIFY	→		→ 200 OK NOTIFY
BYE	←		← BYE 2
200 OK (BYE)	→		→ 200 OK (BYE)

SSS_XXSSCONF_C	NGN reference to:					
RE_06	[20] clause 4.5.2.1.2					
TSS reference:	SIP-SIP/Supplementary_services/ CONF.					
Configuration	CONF					
Selection criteria:	CONF					
Test purpose:	Conference creation by Three-way session creation. REFER request to the focus, Replaces method is used.					
	The conference participant is participating in two SIP sessions and wants to join together two of these active sessions to a so-called three-way session. The replaces method is used to terminate previous individual sessions. The conference participants shall perform the following steps:					
	 Create a conference at the conference focus by sending an INVITE request with the conference factory URI for the three-way session towards the conference focus. Perform for each of the active sessions, that are requested to be joined to the three-way session by performing the procedures for inviting a user to a conference by sending a REFER request to the focus. 					
	The SIP#2 and SIP#3 terminates the session 1 and 2 as indicated in the Replaces header received in the INVTE from the focus					
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT;					
	PIXIT for supported header: Case a) No 100 rel;					
	Case a) No 100 fel;					
	Case b) Supported: 100 rel, Case c) upported: 100 rel and precondition.					
	SIP header values:					
	INVITE: Request URI contained the conference factory URI.					
	200 OK: "isfocus" feature parameter indicated in Contact header field conference URI contained 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-< td=""></sip:user#2?replaces=call-id1%3bto-tagsession1%3bfrom-<>					
	tagSession1; method=INVITE>.					
	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-<="" td=""></sip:user#3?replaces=call-id2%3bto-tag>					
	tag Session2; method=INVITE>. INVITE 4: Replaces=Call-ID1;to-tagSession1:from-tagSession1.					
	BYE 1: Call-ID: call-id1/ To:; tag=session1/ From:;tag=Session1.					
	INVITE 5: Replaces=Call-ID2;to-tag Session2;from-tag Session2.					
	BYE 2: Call-ID: call-id2/ To:; tag=session2/ From:;tag=Session2.					

Comments:		_		217 2		217 // 2
SIP#1		Focus		SIP#2		SIP#3
Establishment of session #						
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		
	Estab	olishment of session #2				
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	È				È	200 OK (INVITE)
ACK	÷				÷	ACK
Conference creation	7				7	ACK
		IND CITE				
INVITE	→	INVITE				
200 OK	←	200 OK				
ACK	→	ACK				
Inviting SIP#2 to the confe	rence	-				
REFER 1	→	REFER				
202 Accepted	←	202 Accepted				
202 / toocpica	•	INVITE 4	→	INVITE		
		200 OK	É	200 OK		
			→			
NOTIFY	,	ACK	7	ACK		
NOTIFY	÷	NOTIFY				
200 OK NOTIFY	→	200 OK NOTIFY	_			
BYE	←		←	BYE 1		
200 OK (BYE)	→		→	200 OK (BYE)		
	Invitir	ng SIP#3 to the conferen	ce			
REFER 2	→	REFER				
202 Accepted	É	202 Accepted				
202 Accepted	~	INVITE 5	_		→	INVITE
			→			
		200 OK	((200 OK
NOTIEV	_	ACK	→		→	ACK
NOTIFY	Ť	NOTIFY				
200 OK NOTIFY	→	200 OK NOTIFY				
BYE	←				←	BYE 2
200 OK (BYE)	→				→	200 OK (BYE)

SSS_XXSSCONF_C	NGN reference to:
RE_07	[20] clause 5.3.1.3.2
TSS reference:	SIP-SIP/Supplementary_services/ CONF.
Configuration	CONF
Selection criteria:	Is not a temporary conference URI sent in the first provisional response when preconditions
	are indicated. Is the use of preconditions supported.
Test purpose:	Conference creation with a conference factory URI. Preconditions indicated a conference
	URI is sent in the first provisional response.
	Ensure that a conference can be created by a UE using the conference factory URI.
	Preconditions are requested by the originating UE. The "isfocus" feature parameter
	indicated in Contact header is received in the 200 OK (INVITE). The conference participant
	shall store the content of the received Contact header as the conference URI.
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
	SIP header values:
	INVITE: Request URI contained the conference factory URI.
	conference URI contained in the Contact header field.
	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.
Comments:	
SIP#1	
INVITE	→
183 Session Progress	(
PRACK	→
200 OK PRACK	(
UPDATE)
200 OK UPDATE	(
200 OK (INVITE)	(
ACK	→

SSSXXSSCONF_C	NGN reference to:			
RE_08	[20] clause 5.3.1.3.3			
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF	F.		
Configuration	CONF			
Selection criteria:	indicated. Is the use of preconditions supported.	st provisional response when preconditions are		
Test purpose:	Conference creation with a conference factor conference URI is sent in the first provisional Ensure that a conference can be created by Preconditions are requested by the originatin Contact header is received in the 200 OK (IN content of the received Contact header as the	I response. a UE using the conference factory URI. ng UE. The "isfocus" feature parameter indicated in IVITE). The conference participant shall store the		
SIP Parameter values:		conference factory URI. contained in the Contact header field. ndicated in Contact header field the Contact header field.		
Comments:				
SIP	P#1 Focu	ıs		
INVITE	→			
183 Session Progress	←			
PRACK	→			
200 OK PRACK	←			
UPDATE	→			
200 OK UPDATE	←			
200 OK (INVITE) ACK	← →			

	NGN reference to: [20] clause 5.3.2.3.1
SSSXXSSCONF_ CRE_09	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.
Configuration	CONF
Selection criteria:	CONF
Test purpose:	Conference creation with a conference factory URI not allocated in the focus unsuccessful.
	Ensure that a conference can not be created by a UE using the conference factory URI not allocated in the focus. The request is rejected by the focus with a 488 Not Acceptable Here final response.
SIP Parameter	Dial string parameters options=PIXIT
values:	TYPE_SDP= PIXIT;
valuoo.	PIXIT for supported header:
	Case a) No 100 rel;
	Case b) Supported: 100 rel;
	Case c) Supported: 100 rel and precondition.
	SIP header values:
	INVITE: Request URI contained the conference factory URI not allocated in the focus
Comments:	
SIF	P#1 Focus
INVITE)
488 Not Acceptable He	
ACK)

SSS_XXSSCONF_C	NGN reference to:
RE_10	[20] clause 5.3.2.3.2
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF
Configuration	CONF
Selection criteria:	CONF
Test purpose:	Conference creation with a conference URI is allocated in the focus unsuccessful.
	Ensure that a conference can not be created by a UE using the conference URI allocated in the focus. The request is rejected by the focus with a 488 Not Acceptable Here final
	response.
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
	SIP header values:
	INVITE: Request URI contained the conference factory URI not allocated in the focus
Comments:	
SIF	P#1 Focus
INVITE	→
488 Not Acceptable Her	re ←
ACK	→

6.5.2.11.2 Joining a conference

SSSXXSSCONF_ Join 01		reference to: clause 5.3.2.4				
TSS reference:		lementary_services/ CONF.				
Configuration	CONF	iomeniary_eerviced/cerii:				
Selection criteria:	CONF					
Test purpose:	Participant dial-in the conference, the conference URI is used. The Participant subscribes the conference event package.					
	received at the co		that conference. An INVITE request is URI is known at the SIP#2. SIP#2 request is successful.			
SIP Parameter		ters options=PIXIT	·			
values:	TYPE_SĎP= PIXI	T;				
	PIXIT for supporte	d header:				
	Case a) No '					
		ported: 100 rel;				
		ported: 100 rel and preconditi				
		equest URI contained the con				
		focus" feature parameter indi				
		nference URI contained in the				
		equest URI contained the con	ference URI, Event contains			
		onference".				
		Event header contains "conference".				
		Y: Event contains conference; Subscription-State contains active; expires=XXXX.				
			contains connected, dialled-in.			
Comments:	αρ	plication/comerence-imo+xim	contains connected, dialied-in.			
SIP#1		Focus	SIP#2			
Conference creation		1 0003	OΠ #2			
INVITE	→	INVITE				
200 OK (INVITE)	-	200 OK (INVITE)				
ACK	→	ACK				
SIP#2 joining in the co	onference					
on "2 joining in the oc	31110101100	INVITE	← INVITE 2			
		18x				
		200 OK INVITE	→ 200 OK INVITE			
		ACK	← ACK			
			← SUBSCRIBE			
			→ 200 OK (SUBSCRIBE)			
		NOTIFY 1	→ NOTIFY`			
		200 OK NOTIFY	← 200 OK NOTIFY			
NOTIFY	←	NOTIFY 2				
200 OK NOTIFY	→	200 OK NOTIFY				

SSSXXSSCONF_	NGN reference to:				
Joint _02	[20] clause 5.3.2.4				
TSS reference:	SIP-SIP/Supplementary_services/ CONF				
Configuration	CONF				
Selection criteria:	It is not possible to invite a participant using a REFER request				
Test purpose:	Participant dial-in the conference, the conference URI is used. The Participant does not subscribe the conference event package.				
	SIP#1 established a conference. SIP#2 joins in that conference. The UE sends an INVITE request to the conferencing AS the conference URI is known at the SIP#2. SIP#2 does not subscribe the conference event package. The request is successful.				
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 SIP header values: INVITE 2: Request URI contained the conference URI. 18x "isfocus" feature parameter indicated in Contact header field. 200 OK: isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field.				
Comments: SIP#1 Conference creation	Focus SIP#2				
INVITE 200 OK (INVITE) ACK	→ INVITE← 200 OK (INVITE)→ ACK				
SIP#2 joining in the co	INVITE ← INVITE 2 18x → 18x 200 OK INVITE → 200 OK INVITE ACK ← ACK				

SSSXXSSCONF_	NGN refer	ence to:				
Join_003	[20] claus	e 5.3.2.4				
TSS reference:	SIP-SIP-SIP/Suppleme	entary_services/ CON	F			
Configuration	CONF					
Selection criteria:	CONF					
Test purpose:	Participant dial-in the o	Participant dial-in the conference, the conference URI is not allocated, the request is rejected.				
				nce. The conference URI in the ejected with the final response 488		
SIP Parameter values:	Dial string parameters TYPE_SDP= PIXIT; PIXIT for supported he Case a) No 100 r Case b) Supporte Case c) Supporte SIP header values: INVITE 2: Request Uf	eader: rel ed: 100 rel ed: 100 rel and precon		llocated in the focus		
Comments: SIP#1		Focus		SIP#2		
Conference creation INVITE 200 OK (INVITE) ACK	→ ← →	INVITE 200 OK (INVITE) ACK				
SIP#2 joining in the conf	ierence	488 Not Accep		INVITE 2 488 Not Acceptable Here ACK		

6.5.2.11.3 Inviting other users to a conference

SSS_XXSSCONF_I NV_001		GN reference to: D] clause 5.3.2.5	
TSS reference:		Supplementary_services/ CON	I. IF
Configuration	CONF		·
Selection criteria:	CONF		
Test purpose:	Inviting partici subscribed.	pant by sending REFER to the	e focus. The conference event package is
	SIP#1 establis	shed a conference. SIP#1 invit	tes SIP#2 to join into the conference. The SIP#1
	sends a REFE		ds an INVITE request to SIP#2 to invite the SIP#2 to
SIP Parameter values:			
	TYPE_SDP= I		
	PIXIT for supp		
		No 100 rel	
	,	Supported: 100 rel	
		Supported: 100 rel and precor	ndition
	SIP header va		
	REFER:	Request URI contained the c	onference URI.
		Refer-To contains the URI of	SIP#2, method=invite.
		Referred-By contains SIP or	tel URI of SIP#1.
	INVITE 2:	Request URI contained the c	onference URI.
		The P-Asserted-Identity conta	ains the conference URI.
		"isfocus" feature parameter ir	ndicated in Contact header field
		conference URI contained in	the Contact header field.
		Referred-By contains SIP or	tel URI of SIP#1.
	NOTIFY 1	Event contains conference ; message/sipfrag contains SII	Subscription-State contains active P/2.0 100 Trying.
	NOTIFY 2		Subscription-State contains active
	SUBSCRIBE:		onference URI, Event contains "conference".
	NOTIFY 3		Subscription-State contains active; expires=XXXX
			rml contains connected, dialled-out.
	NOTIFY 4		Subscription-State contains active
			rml contains connected, dialled-out.
		Continued	

Comments: SIP#1 SIP#2 **Focus** Conference creation INVITE **→** INVITE 200 OK (INVITE) 200 OK (INVITE) ACK **ACK** SUBSCRIBE 200 OK (SUBSCRIBE) SIP#1 invites SIP#2 to the conference REFER **→ REFER** 202 Accepted 202 Accepted Focus dials out to invite SIP#2 INVITE 2 → INVITE NOTIFY **NOTIFY 1 ←** 200 OK NOTIFY 200 OK NOTIFY 180 Ringing **←** 180 Ringing 200 OK INVITE **←** 200 OK INVITE **ACK** ACK **→** NOTIFY NOTIFY 2 200 OK NOTIFY **→** 200 OK NOTIFY **SUBSCRIBE →** 200 OK (SUBSCRIBE) **NOTIFY →** NOTIFY 3 200 OK NOTIFY 200 OK NOTIFY **← NOTIFY 4** NOTIFY 200 OK NOTIFY 200 OK NOTIFY

SSS_XXSSCONF_I NV_002	NGN refe [20] claus			
TSS reference:	SIP-SIP-SIP/Supplen	nentary_services/ CONI	=.	
Configuration	CONF			
Selection criteria:	CONF			
Test purpose:	Inviting participant by subscribed.	sending REFER to the	focus. Th	e conference event package is not
	sends a REFER to th		s an INVI7	o join into the conference. The SIP#1 FE request to SIP#2 to invite the SIP#2 to subscribed.
SIP Parameter values:			9	
	TYPE_SDP= PIXIT;			
	PIXIT for supported h	neader:		
	Case a) No 100			
	Case b) Suppor	ted: 100 rel;		
	Case c) Suppor	ted: 100 rel and precon	dition.	
	SIP header values:			
		t URI contained the con		
		o contains the URI of S		nod=invite.
		d-By contains SIP URI		
		t URI contained the con		
		Asserted-Identity contain		
		" feature parameter ind		
		nce URI contained in th		
		d-By contains SIP or tel		
		ontains conference; Sub		
		e/sipfrag contains SIP/2 ontains conference; Sul		
		e/sipfrag contains SIP/2		
1				
Comments:	messag	e/sipirag contains on /2	0 200 01	· ·
Comments:		<u> </u>	200 01	
SIP#1	messag	Focus	<u> </u>	SIP#2
SIP#1 Conference creation	_	Focus	200 01	
SIP#1 Conference creation INVITE	→ +	Focus	200 01	
SIP#1 Conference creation	→	Focus	200 01	
SIP#1 Conference creation INVITE 200 OK (INVITE)	→ ← →	Focus INVITE 200 OK (INVITE)	200 01	
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK	→ ← →	Focus INVITE 200 OK (INVITE)	200 OI	
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SIP#1 invites SIP#2 to	+ + + + + + + + + + + + + + + + + + +	Focus INVITE 200 OK (INVITE) ACK	<u> 200 O1</u>	
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SIP#1 invites SIP#2 to REFER	→ the conference	Focus INVITE 200 OK (INVITE) ACK REFER 202 Accepted		SIP#2
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SIP#1 invites SIP#2 to REFER	→ the conference	Focus INVITE 200 OK (INVITE) ACK REFER 202 Accepted	Focus dials	SIP#2 s out to invite SIP#2
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SIP#1 invites SIP#2 to REFER 202 Accepted	the conference	Focus INVITE 200 OK (INVITE) ACK REFER 202 Accepted INVITE	Focus dials	SIP#2
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SIP#1 invites SIP#2 to REFER 202 Accepted	the conference	Focus INVITE 200 OK (INVITE) ACK REFER 202 Accepted INVIT	Focus dials	SIP#2 s out to invite SIP#2
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SIP#1 invites SIP#2 to REFER 202 Accepted	the conference	Focus INVITE 200 OK (INVITE) ACK REFER 202 Accepted INVIT NOTIFY 1 200 OK NOTIFY	Focus dials E 2 →	SIP#2 s out to invite SIP#2 INVITE
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SIP#1 invites SIP#2 to REFER 202 Accepted	the conference	Focus INVITE 200 OK (INVITE) ACK REFER 202 Accepted FINVIT NOTIFY 1 200 OK NOTIFY 180 Ring	Focus dials E 2 →	SIP#2 s out to invite SIP#2 INVITE 180 Ringing
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SIP#1 invites SIP#2 to REFER 202 Accepted	the conference	Focus INVITE 200 OK (INVITE) ACK REFER 202 Accepted INVIT NOTIFY 1 200 OK NOTIFY 180 Ring 200 OK INV	Focus dials E 2 → ging ← ITE ←	SIP#2 s out to invite SIP#2 INVITE 180 Ringing 200 OK INVITE
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SIP#1 invites SIP#2 to REFER 202 Accepted NOTIFY 200 OK NOTIFY	the conference	Focus INVITE 200 OK (INVITE) ACK REFER 202 Accepted INVIT NOTIFY 1 200 OK NOTIFY 180 Ring 200 OK INV	Focus dials E 2 →	SIP#2 s out to invite SIP#2 INVITE 180 Ringing
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SIP#1 invites SIP#2 to REFER 202 Accepted	the conference	Focus INVITE 200 OK (INVITE) ACK REFER 202 Accepted INVIT NOTIFY 1 200 OK NOTIFY 180 Ring 200 OK INV	Focus dials E 2 → ging ← ITE ←	SIP#2 s out to invite SIP#2 INVITE 180 Ringing 200 OK INVITE

SSS_XXSSCONF_	N	IGN reference to:				
INV_003	[2	20] clause 5.3.2.5				
TSS reference:	SIP-SIP-SIF	P/Supplementary_services/ CC	NF.			
Configuration	CONF					
Selection criteria:	CONF					
Test purpose:	Inviting part subscribed.	icipant by sending REFER to	the participant. The conference event package is			
	a REFER to		vites SIP#2 to join into the conference by sending nt sends an INVITE request to the focus to dial in.			
SIP Parameter		arameters options=PIXIT				
values:	TYPE_SDP	= PIXIT;				
	PIXIT for su	pported header:				
	Case a)					
	Case b)					
	Case c)		condition.			
	SIP header					
	REFER:	Request URI contained the U				
		Refer-To contains the confer				
		Referred-By contains the UR				
	INVITE 2:	Request URI containes the c				
		The P-Asserted-Identity cont				
			ndicated in Contact header field			
		conference URI contained in				
	NOTIEN	Referred-By contains SIP or				
	NOTIFY 1		Subscription-State contains active.			
			Subscription-State contains active			
	NOTIEVA	message/sipfrag contains SI				
	NOTIFY 2		Subscription-State contains active.			
			Subscription-State contains active			
	CLIDCCDID	message/sipfrag contains SI				
	NOTIFY 3	SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference ; Subscription-State contains active ;				
	NOTH 13	expires=XXXX				
			kml contains connected, dialled-in.			
	NOTIFY 4		Subscription-State contains active			
			kml contains connected, dialled-in.			
		Continued				

Comments: SIP#1	Focus	SIP#2
Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) SIP#1 invites SIP#2 to the conference REFER	INVITE 200 OK (INVITE) ACK	→ REFER
202 Accepted		€ 202 Accepted
NOTIFY 200 OK NOTIFY NOTIFY 200 OK NOTIFY	SIP#2 INVITE 180 Ringing 200 OK INVITE ACK	 NOTIFY 1 → 200 OK NOTIFY joins in the conference ← INVITE 2 → 180 Ringing → 200 OK INVITE ← ACK ← NOTIFY 2 → 200 OK NOTIFY
NOTIFY 200 OK NOTIFY		

SSS_XXSSCONF_I NV_004		ference to: use 5.3.2.5		
TSS reference:	SIP-SIP-SIP/Sup	plementary_services/	CONF.	
Configuration	CONF			
Selection criteria:	CONF			
	Inviting participan not subscribed.	nt by sending REFER	to the particip	pant. The conference event package is
	sending a REFER		ne participant	2 to join into the conference by sends an INVITE request to the focus scribed.
SIP Parameter values:			,0 .001 00.00	
	TYPE_SDP= PIX			
	PIXIT for support			
	Case a) No			
	Case b) Sup	ported: 100 rel;		
	Case c) Sup	ported: 100 rel and pr	recondition.	
	SIP header value			
		quest URI contained t		
		fer-To contains the co		
		ferred-By contains the		
		quest URI contained the		
		e P-Asserted-Identity of		
				in Contact header field
		ference URI containe		
		ferred-By contains SIF		
				tion-State contains active
		ssage/sipfrag contains		
		ssage/sipfrag contains		tion-State contains active
Comments:	IIIC	33age/3ipirag contains	3 311 72.0 200	OK.
SIP#1		Focus		SIP#2
Conference creation		1 0003		Oli #2
INVITE	→	INVITE		
200 OK (INVITE)	÷	200 OK (INVITE)		
ACK	→	ACK		
SIP#1 invites SIP#2 to	the conference	,		
REFER	→		→	REFER
202 Accepted	←		←	202 Accepted
NOTIFY	←		←	NOTIFY 1
200 OK NOTIFY	→		→	200 OK NOTIFY
			SIP#2 joins	in the conference
			√IVITE ←	INVITE 2
		400 D	inging 👈	400 Dinging
		180 Ri		180 Ringing
		180 RI 200 OK IN	IVITĔ →	200 OK INVITE
		200 OK IN		
NOTIFY 200 OK NOTIFY	←		IVITĔ →	200 OK INVITE

TSS reference: SIP-SIP-SIP/Supplementary_services/ CONF. Configuration CONF Test purpose: Focus invites to conference by sending REFER to the participant. The conference event package is subscribed. SIP#1 established a conference. The focus invites SIP#2 to join into the conference by sending a REFER to this participant. The participant sends an INVITE request to the foc dial in. The conference event package is subscribed. SIP Parameter values: Dial string parameters options=PINIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case b) Supported: 100 rel and precondition. SIP header values: REFER: Request URI contained the URI of SIP#2. Refer-To contains the conference URI, method=invite. The P-Asserted-Identity contains the conference URI. INVITE 2: Request URI containes the conference URI. INVITE 2: Request URI containes the conference URI. INVITE 2: Request URI containes the conference URI. The P-Asserted-Identity contains the URI of SIP#2. "Isfocus" feature parameter indicated in Contact header field conference URI contained the URI of SIP#2. Refer-To contained the Contact header field. NOTIFY 1: Event contains conference; Subscription-State contains active message/sipfrag contains SIP#2.0 100 Trying. NOTIFY 2: Event contains conference; Subscription-State contains active message/sipfrag contains SIP#2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains active message/sipfrag contains SIP#2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains active application/conference-info+xml contains connected, dialled-in. NOTIFY 3: Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. NOTIFY 4: Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4: 200 OK (INVITE) ACK	SSS_XXSSCONF_I NV_005		erence to: use 5.3.2.5		
Configuration CONF				NF.	
Selection criteria: CONF			omentary_convictor oc		
Test purpose: Focus Invites to conference by sending REFER to the participant. The conference event package is subscribed. SIP#1 established a conference. The focus invites SIP#2 to join into the conference by sending a REFER to this participant. The participant sends an INVITE request to the foot dial in. The conference event package is subscribed. This applicable by a web interface. SIP Parameter					
package is subscribed. SIP#1 established a conference. The focus invites SIP#2 to join into the conference by sending a REFER to this participant. The participant sends an INVITE request to the focidal in. The conference event package is subscribed. This applicable by a web interface. SIP Parameter values:			nference by sending RI	FFR to the participant. Th	e conference event
sending a REFER to this participant. The participant sends an INVITE request to the foot dial in. The conference event package is subscribed. This applicable by a web interface. 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 and precondition. SIP header values: REFER: Request URI contained the URI of SIP#2. Refer-To contains the conference URI, method=invite. The P-Asserted-Identity contains the conference URI. INVITE 2: Request URI contained the URI of SIP#2. "Isfocus" feature parameter indicated in Contact header field conference URI contains the conference URI. NOTIFY 1: Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2: Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 3: Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3: Event contains conference; Subscription-State contains active expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4: Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. NOTIFY 4: Vent contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. NOTIFY 4: NOTIFY 4: NOTIFY 1: 200 OK NOTIFY 2: NOTIFY 1: 200 OK NOTIFY 3: 200 OK INVITE 4: 200 OK INVITE 4: 200 OK INVITE 3: 200 OK INVITE 4: 200 OK INV	rest parpess.		, ,		
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 and precondition. SIP header values: REFER: Request URI contained the URI of SIP#2. Refer-To contains the conference URI, method=invite. The P-Asserted-Identity contains the conference URI. INVITE 2: Request URI containes the conference URI. INVITE 2: Request URI contained in the URI of SIP#2. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field conference URI contains stellow message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 1 Event contains conference, Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains active application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference: Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Pour Invite		sending a REFER to	o this participant. The լ	participant sends an INVIT	E request to the focus to
values: TYPE_SDP= PIXIT; PIXIT for supported header:	CID Doromotor			ubscribed. This applicable	by a web interface.
PIXIT for supported header:					
Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition. SIP header values: REFER: Request URI contained the URI of SIP#2. Refer-To contains the conference URI, method=invite. The P-Asserted-Identity contains the conference URI. INVITE 2: Request URI containes the conference URI. INVITE 2: Request URI containes the conference URI. The P-Asserted-Identity contains the URI of SIP#2. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 00 OK. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 5 INVITE 5 ACK SUBSCRIBE 7 REFER 200 OK (INVITE) ACK SUBSCRIBE 9 ACK 100 OK (INVITE) ACK 100 OK (INVI	values.				
Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition. SIP header values: REFER: Request URI contained the URI of SIP#2. Refer-To contains the conference URI, method=invite. The P-Asserted-Identity contains the conference URI. INVITE 2: Request URI containes the conference URI. The P-Asserted-Identity contains the URI of SIP#2. "isfocus" feature parameter indicated in Contact header field. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE Pocus SIP#2 Focus SIP#2 Focus Focus Focus SIP#2 to the conference REFER REFER 202 Accepted NOTIFY 2 200 OK NOTIFY SIP#2 joins in the conference INVITE 4 NOTIFY 2 200 OK NOTIFY SIP#2 joins in the conference INVITE 100 OK NOTIFY 2 200 OK NOTIFY SIP#2 joins in the conference INVITE 4 NOTIFY 4 NOTIFY 2					
Case c) SIP header values: REFER: REFER: REQUest URI contained the URI of SIP#2. Refer-To contains the conference URI, method=invite. The P-Asserted-Identity contains the conference URI. INVITE 2: Request URI containes the conference URI. The P-Asserted-Identity contains the URI of SIP#2. "isfocus" feature parameter indicated in Contact header field conference URI containes on ference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 0K. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 0K. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 4 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Conference creation INVITE 200 OK (INVITE) ACK		,	,		
SIP header values: REFER: Request URI contained the URI of SIP#2. Refer-To contains the conference URI, method=invite. The P-Asserted-Identity contains the conference URI. INVITE 2: Request URI containes the conference URI. The P-Asserted-Identity contains the URI of SIP#2. "Isfocus" feature parameter indicated in Contact header field conference URI contains the URI of SIP#2. "Isfocus" feature parameter indicated in Contact header field conference URI contains the URI of SIP#2. "Isfocus" feature parameter indicated in Contact header field conference URI contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. Portion of the vent contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Conference creation INVITE ACK → ACK → ACK → ACK SUBSCRIBE Focus SIP#2 to the conference REFER 202 Accepted ← 202 Accepted NOTIFY ← NOTIFY 1 200 OK (SUBSCRIBE) Focus invites SIP#2 to the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2				ondition.	
Refer-To contains the conference URI, method=invite. The P-Asserted-Identity contains the conference URI. INVITE 2: Request URI containes the conference URI. The P-Asserted-Identity contains the URI of SIP#2. "isfocus" feature parameter indicated in Contact header field conference URI contains dender field. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 4 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Pocus SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE ** Focus ** SIP#2 ** INVITE ** OO OK (INVITE) ACK ** ACK ** ACK ** ONTIFY ** NOTIFY 1 ** 200 OK NOTIFY ** 200 OK NOTIFY ** SIP#2 joins in the conference INVITE ** INVITE 2 ** 180 Ringing ** 180 Ringing ** 200 OK INVITE ** ACK ** ACK ** NOTIFY ** NOTIFY 2					
The P-Asserted-Identity contains the conference URI. INVITE 2: Request URI containes the conference URI. The P-Asserted-Identity contains the URI of SIP#2. "isfocus" feature parameter indicated in Contact header field conference URI contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (INVITE) ACK SUBSCRIBE Focus SIP#2 Focus SIP#2 Focus SIP#2 Conference Creation INVITE ACK SUBSCRIBE POUS NOTIFY + NOTIFY 1 200 OK NOTIFY - 200 OK NOTIFY SIP#2 joins in the conference INVITE - 180 Ringing - 180 Ringing 200 OK INVITE - 200 OK INVITE ACK - ACK NOTIFY - NOTIFY 2		REFER: Reque	st URI contained the U	RI of SIP#2.	
INVITE 2: Request URI containes the conference URI.					
The P-Asserted-Identity contains the URI of SIP#2. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Conference creation INVITE					
"isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Focus SIP#2 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE → INVITE 200 OK (SUBSCRIBE) Focus invites SIP#2 to the conference REFER → REFER 202 Accepted ← 202 Accepted NOTIFY ← NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE → 200 OK INVITE ← ACK ← ACK NOTIFY ← NOTIFY 2					
conference URI contained in the Contact header field. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) Focus SIP#2 Focus SIP#2 Focus Focus invites SIP#2 to the conference REFER Conference Conference REFER REFER 202 Accepted NOTIFY NOTIFY 200 OK NOTIFY SIP#2 joins in the conference INVITE INVITE 180 Ringing 200 OK INVITE 2180 Ringing 200 OK INVITE ACK ACK ACK ACK NOTIFY NOTIFY NOTIFY 2					field.
NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) Focus SIP#2 INVITE 200 OK (INVITE) ACK SUBSCRIBE ↑ Focus invites SIP#2 to the conference REFER 202 Accepted NOTIFY ← NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2					Tiela
message/sipfrag contains SIP/2.0 100 Trying. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Focus SIP#2 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) Focus invites SIP#2 to the conference REFER → REFER 202 Accepted ← 202 Accepted NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK IN					active
message/sipfrag contains SIP/2.0 200 OK. SUBSCRIBE: Request URI contained the conference URI, Event contains "conference". NOTIFY 3 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Focus SIP#2 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) Focus SIP#2 Focus SIP#2 Focus SIP#2 Focus invites SIP#2 to the conference REFER REFER 202 Accepted 202 Accepted NOTIFY NOTIFY NOTIFY 1 200 OK NOTIFY NOTIFY 2 180 Ringing 180 Ringing 200 OK INVITE 2 180 Ringing 180 Ringing 200 OK INVITE 2 200 OK INVITE 3 200 OK INVITE 3 200 OK INVITE 4 ACK ACK NOTIFY NOTIFY 2		messa	ge/sipfrag contains SIF	/2.0 100 Trying.	
NOTIFY 3 Event contains conference; Subscription-State contains active; expires=XXXX application/conference-info+xml contains connected, dialled-in. NOTIFY 4 Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Focus SIP#2 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) Focus SIP#2 Focus Focus invites SIP#2 to the conference REFER → REFER 202 Accepted ← 202 Accepted NOTIFY 1 200 OK NOTIFY ← NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2					
expires=XXXX application/conference-info+xml contains connected, dialled-in. Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) Focus SIP#2 INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) Focus invites SIP#2 to the conference REFER 202 Accepted ← 202 Accepted NOTIFY ← NOTIFY 1 200 OK NOTIFY ← NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2					
application/conference-info+xml contains connected, dialled-in. Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) Focus INVITE				Subscription-State contains	active;
NOTIFY 4 Event contains conference; Subscription-State contains active application/conference-info+xml contains connected, dialled-in. Comments: SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) focus invites SIP#2 to the conference REFER PREFER 202 Accepted POTIFY NOTIFY NOTIFY 1 200 OK NOTIFY NOTIFY 2 180 Ringing POOK INVITE ACK NOTIFY ACK NOTIFY COOK INVITE ACK ACK NOTIFY ACK NOTIFY NOTIFY 2					
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Comments: SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) Focus F					
SIP#1 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) focus invites SIP#2 to the conference REFER → REFER 202 Accepted ← 202 Accepted NOTIFY ← NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2	Comments:	аррііса	ation/contenence-inio+x	ini contains connected, die	aneu-iii.
INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) focus invites SIP#2 to the conference REFER REFER 202 Accepted 202 Accepted NOTIFY NOTIFY NOTIFY 1 200 OK NOTIFY NOTIFY SIP#2 joins in the conference INVITE INVITE 1 180 Ringing 180 Ringing 200 OK INVITE ACK ACK NOTIFY NOTIFY 2			Focus		SIP#2
200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) focus invites SIP#2 to the conference REFER → REFER 202 Accepted ← 202 Accepted NOTIFY ← NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2	Conference creation				
ACK SUBSCRIBE 200 OK (SUBSCRIBE) focus invites SIP#2 to the conference REFER → REFER 202 Accepted ← 202 Accepted NOTIFY ← NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2	INVITE	→	INVITE		
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focus invites SIP#2 to the conference REFER → REFER 202 Accepted ← 202 Accepted NOTIFY ← NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2	SUBSCRIBE	→			
REFER → REFER 202 Accepted ← 202 Accepted NOTIFY ← NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2	200 OK (SUBSCRIBE)	←			
202 Accepted NOTIFY ← NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2					erence
NOTIFY ← NOTIFY 1 200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2					J
200 OK NOTIFY → 200 OK NOTIFY SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2					ı
SIP#2 joins in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2					IFY
INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2					
180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2				•	100
200 OK INVITE → 200 OK INVITE ACK ← ACK NOTIFY ← NOTIFY 2					
NOTIFY ← NOTIFY 2					ΓE
200 OK NOTIFY → 200 OK NOTIFY					
			200 OK NO	IIFY → 200 OK NOT	l⊦Y
∠ CURCODIRE				∠ CURCORISE	
← SUBSCRIBE→ 200 OK (SUBSCRIBE)					SCDIRE)
NOTIFY 3 → NOTIFY			NOTI		JOONIDL)
200 OK NOTIFY C 200 OK NOTIFY					IFY
NOTIFY • NOTIFY 4	NOTIFY	←		= 200 010101	:- :
200 OK NOTIFY → 200 OK NOTIFY	_				

SSS_XXSSCONF_ INV_00		erence to: se 5.3.2.5		
TSS reference:		mentary_services/ CONF.		
Configuration	CONF	•		
Selection criteria:	CONF			
Test purpose:	Focus invites by ser subscribed.	nding REFER to the participa	ant. The	conference event package is
	sending a REFER to		pant sen	2 to join into the conference by ds an INVITE request to the focus to
SIP Parameter	Dial string paramete			
values:	TYPE_SDP= PIXIT;			
	PIXIT for supported			
	Case a) No 10			
	Case b) Suppo			
		orted: 100 rel and precondition	on.	
	SIP header values: REFER: Reques	st URI contained the URI of \$	CID#2	
		o contains the conference		thod-invite
		Asserted-Identity contains th		
		at URI containes the confere		oneo ora:
		Asserted-Identity contains th		SIP#2.
		s" feature parameter indicate		
		nce URI contained in the Co		
		contains conference ; Subsc		
		ge/sipfrag contains SIP/2.0 1		
		contains conference; Subsci		tate contains active
Comments:	messaç	ge/sipfrag contains SIP/2.0 2	200 OK	
SIP#1		Focus		SIP#2
Conference creation		1 0003		011 #Z
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK ` ´	→	ACK `		
SUBSCRIBE	→			
200 OK (SUBSCRIBE)	(
				2 to the conference
		REFER		REFER
		202 Accepted		202 Accepted
		NOTIFY		_
		200 OK NOTIFY	→ 2	200 OK NOTIFY
		SID#3	2 ining in	the conference
		INVITE		NVITE 2
		180 Ringing		80 Ringing
		200 OK INVITE	-	200 OK INVITE
		ACK		ACK
		NOTIFY	← N	NOTIFY 2
		200 OK NOTIFY	→ 2	200 OK NOTIFY

6.5.2.11.4 Leaving a conference

SSS_XXSSCONF	NGN refe					
LEAV_001	[20] claus					
TSS reference:		mentary_services/ COI	NF			
Configuration	CONF					
Selection criteria:	CONF	 				
Test purpose:	A participant leaves the conference. The conference event package is subscribed.					
	SIP#2 wishes to leave the conference by sending a BY request to the focus in accordance of the basic call procedures. The conference event package is subscribed.					
SIP Parameter		Dial string parameters options=PIXIT				
values:	TYPE_SDP= PIXIT;					
	PIXIT for supported h					
	Case a) No 100					
		rted: 100 rel;	11.01			
		rted: 100 rel and preco	indition.			
	SIP header values:	ntaina aamf		. Chata agentaina tammalasata d		
				n-State contains terminated		
	xml document: <user></user> uri=SIP#2, status is departed .					
	NOTIFY 2 Event contains conference ; Subscription-State contains active xml document: <user></user> uri=SIP#2, status is departed .					
0 1	xmi docu	ment: <user></user> uri=5iPi	#Z, Status	is departed.		
Comments: SIP#1		Focus		SIP#2		
Conference creation		rocus		31F#2		
INVITE	→	INVITE				
200 OK (INVITE)	÷	200 OK (INVITE)				
ACK	→	ACK				
SUBSCRIBE	→	AON				
200 OK (SUBSCRIBE	=					
SIP#2 joining in the o						
	Silierence	IN	VITE €	INVITE 2		
		180 Rir		180 Ringing		
		200 OK IN		200 OK INVITE		
				- ACK		
Conference communi	caton		AON •	AOR		
	oaton					
Participant leaves the	converence					
. a. noipain loavoo tilo	23.110101100		•	- BYE		
			-			
			-			
		NOTI	FY 1 -	NOTIFY		
		200 NO				
NOTIFY	←	NOTIFY 2				
200 NOTIFY	→	200 NOTIFY				
l						

LEAV_002	SSS_XXSSCONF	NGN refe				
Configuration Selection criteria: CONF Test purpose: A participant leaves the conference. The conference event package is not subscribed. SIP#2 wishes to leave the conference by sending a BY request to the focus in accordance to the basic call procedures. The conference event package is not subscribed. SIP Parameter values: 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#1 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) SIP#2 joining in the conference INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE ACK ACK Conference communicaton Participant leaves the converence EBYE	_LEAV_002					
Selection criteria: CONF Test purpose: A participant leaves the conference. The conference event package is not subscribed. SIP#2 wishes to leave the conference by sending a BY request to the focus in accordance to the basic call procedures. The conference event package is not subscribed. 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#1 Focus SIP#2 Conference creation INVITE → INVITE → ACK SUBSCRIBE ⇒ ACK SIP#2 joining in the conference INVITE ← INVITE 2 180 Ringing 200 OK INVITE → 200 OK INVITE ACK Conference communication Participant leaves the converence ← BYE			nentary_services/ CON	NF.		
Test purpose: A participant leaves the conference. The conference event package is not subscribed. SIP#2 wishes to leave the conference by sending a BY request to the focus in accordance to the basic call procedures. The conference event package is not subscribed. 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 b) Supported: 100 rel and precondition. Comments: SIP#1 Focus SIP#2 Conference creation INVITE → INVITE → ACK → ACK → ACK SIP#2 joining in the conference INVITE 180 Ringing → 180 Ringing 200 OK INVITE → ACK Conference communicaton Participant leaves the converence FYE						
SIP#2 wishes to leave the conference by sending a BY request to the focus in accordance to the basic call procedures. The conference event package is not subscribed. SIP Parameter Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header:	Selection criteria:					
SIP Parameter values: Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header:	Test purpose:	SIP#2 wishes to leav	e the conference by se	ending	a BY	request to the focus in accordance
values: TYPE_SDP= PIXIT; PIXIT for supported header:				e even	it pac	kage is not subscribed.
Comments: SIP#1 Focus SIP#2 Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) SIP#2 joining in the conference INVITE 180 Ringing 200 OK INVITE ACK Conference communication Participant leaves the converence Focus SIP#2 INVITE 200 OK (INVITE) ACK ACK BYE		TYPE_SDP= PIXIT; PIXIT for supported h Case a) No 100 Case b) Suppor	neader: o rel; rted: 100 rel;	ndition		
Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) SIP#2 joining in the conference INVITE 180 Ringing 200 OK INVITE ACK ACK Conference communication Participant leaves the converence **Prescription* Page 180 Ringing Ack ACK ACK BYE	Comments:	, ,	•			
INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE) SIP#2 joining in the conference INVITE 100 OK (INVITE) ACK HIVITE 100 OK (INVITE) AC	SIP#1		Focus			SIP#2
INVITE ← INVITE 2 180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK Conference communication Participant leaves the converence ← BYE	INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE	← → → → ←	200 OK (INVITE)			
Participant leaves the converence ← BYE			180 Rir 200 OK IN'	nging VITE	→	180 Ringing 200 OK INVITE
← BYE	Conference communication	caton			=	
- · -	Participant leaves the	converence			4	RVE
					_	

TSS reference: SIP-SIP/Supplementary_services/ CONF. Configuration CONF Selection criteria: CONF Test purpose: A participant wishes to remove an other participant from the conference. The confere event package is subscribed. SIP#1 wishes to remove SIP#2 from the conference. SIP#2 receives a REFER reque SIP#1 contains the conference URI and the method value BYE in the Refer-To heade SIP#2 sends a BYE to the focus. The conference event package is subscribed. SIP Parameter 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. SIP header values: REFER 1: Request URI contained the URI of SIP#2 URI. Refer-To contains the Conference URI, method=BYE. Referred-By contains the URI of SIP#1 URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xmll document: <user>uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains terminated xmll document: <user>uri=SIP#2, status is departed.</user></user>	st from
Configuration Selection criteria: CONF Selection criteria: CONF Test purpose: A participant wishes to remove an other participant from the conference. The conference event package is subscribed. SIP#1 wishes to remove SIP#2 from the conference. SIP#2 receives a REFER reque SIP#1 contains the conference URI and the method value BYE in the Refer-To heade SIP#2 sends a BYE to the focus. The conference event package is subscribed. SIP Parameter 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. SIP header values: REFER 1: Request URI contained the URI of SIP#2 URI. Refer-To contains the conference URI, method=BYE. Referred-By contains the URI of SIP#1 URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <u>users uri=SIP#2, status is departed.</u> NOTIFY 4 Event contains conference; Subscription-State contains active	st from
Selection criteria: Test purpose: A participant wishes to remove an other participant from the conference. The confere event package is subscribed. SIP#1 wishes to remove SIP#2 from the conference. SIP#2 receives a REFER reque SIP#1 contains the conference URI and the method value BYE in the Refer-To heade SIP#2 sends a BYE to the focus. The conference event package is subscribed. 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; and precondition. SIP header values: REFER 1: Request URI contained the URI of SIP#2 URI. Refer-To contains the conference URI, method=BYE. Referred-By contains the URI of SIP#1 URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <u 4="" active<="" conference;="" contains="" departed.="" event="" is="" notify="" serve="" sip#2,="" status="" subscription-state="" td="" uni="" ≤=""><td>st from</td></u>	st from
Test purpose: A participant wishes to remove an other participant from the conference. The conference event package is subscribed. SIP#1 wishes to remove SIP#2 from the conference. SIP#2 receives a REFER reque SIP#1 contains the conference URI and the method value BYE in the Refer-To heade SIP#2 sends a BYE to the focus. The conference event package is subscribed. 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. SIP header values: REFER 1: Request URI contained the URI of SIP#2 URI. Refer-To contains the Conference URI, method=BYE. Referred-By contains the URI of SIP#1 URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xmll document: <use>user>uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</use>	st from
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SIP#1 contains the conference URI and the method value BYE in the Refer-To heade SIP#2 sends a BYE to the focus. The conference event package is subscribed. 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. SIP header values: REFER 1: Request URI contained the URI of SIP#2 URI. Refer-To contains the conference URI, method=BYE. Referred-By contains the URI of SIP#1 URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <user> user> uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</user>	
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. SIP header values: REFER 1: Request URI contained the URI of SIP#2 URI. Refer-To contains the conference URI, method=BYE. Referred-By contains the URI of SIP#1 URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <user> uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</user>	
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Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition. SIP header values: REFER 1: Request URI contained the URI of SIP#2 URI. Refer-To contains the conference URI, method=BYE. Referred-By contains the URI of SIP#1 URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <use>use uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</use>	
Case c) Supported: 100 rel and precondition. SIP header values: REFER 1: Request URI contained the URI of SIP#2 URI. Refer-To contains the conference URI, method=BYE. Referred-By contains the URI of SIP#1 URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <use>use viri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</use>	
SIP header values: REFER 1: Request URI contained the URI of SIP#2 URI. Refer-To contains the conference URI, method=BYE. Referred-By contains the URI of SIP#1 URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <user> uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</user>	
Refer-To contains the conference URI, method=BYE. Referred-By contains the URI of SIP#1 URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <user>uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</user>	
Referred-By contains the URI of SIP#1 URI. NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <user> uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</user>	
NOTIFY 1 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <user> uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</user>	
message/sipfrag contains SIP/2.0 100 Trying. The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <user> uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</user>	
The P-Asserted-Identity contains the conference URI. NOTIFY 2 Event contains conference ; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference ; Subscription-State contains terminated xml document: <user></user> uri=SIP#2, status is departed. NOTIFY 4 Event contains conference ; Subscription-State contains active	
NOTIFY 2 Event contains conference; Subscription-State contains active message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <user> uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</user>	
message/sipfrag contains SIP/2.0 200 OK. The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference; Subscription-State contains terminated xml document: <user> uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</user>	
The P-Asserted-Identity contains the conference URI. NOTIFY 3 Event contains conference ; Subscription-State contains terminated xml document: <user></user> uri=SIP#2, status is departed . NOTIFY 4 Event contains conference ; Subscription-State contains active	
NOTIFY 3 Event contains conference ; Subscription-State contains terminated xml document: <user></user> uri=SIP#2, status is departed . NOTIFY 4 Event contains conference ; Subscription-State contains active	
xml document: <user> uri=SIP#2, status is departed. NOTIFY 4 Event contains conference; Subscription-State contains active</user>	
NOTIFY 4 Event contains conference; Subscription-State contains active	
xml document: <user> uri=SIP#2 status is departed</user>	
Ann doddinont. Substruction #2, status is departed.	
Comments:	
SIP#1 Focus SIP#2	
Conference creation INVITE → INVITE	
INVITE 200 OK (INVITE) → INVITE 200 OK (INVITE)	
ACK → ACK	
SUBSCRIBE +	
200 OK (SUBSCRIBE)	
SIP#2 joining in the conference	
INVITE ← INVITE 2	
180 Ringing → 180 Ringing 200 OK INVITE → 200 OK INVITE ACK ← ACK	
Conference communication	
SIP#1 wishes to remove SIP#2 from the conference	
REFER 1 → REFER	
202 Accepted ← 202 Accepted	
NOTIFY NOTIFY 1	
200 OK NOTIFY → 200 OK NOTIFY	
₹ DVE	
 ♣ BYE → 200 OK BYE 	
NOTIFY NOTIFY NOTIFY 2	
200 OK NOTIFY → 200 OK NOTIFY	
200 01(101111	
NOTIFY 3 → NOTIFY	
200 NOTIFY ← 200 NOTIFY	
NOTIFY ← NOTIFY 4	
200 NOTIFY → 200 NOTIFY	

LEAV_004					
TSS reference:		ause 5.3.2.6 pplementary_services/ CON	IE		
	CONF	ppierrieritary_services/ CON	NI .		
	CONF				
		hes to remove an other nar	ticinant	from the conference. The conference	
	event package is		истрат	nom the conterence. The conterence	
	from SIP#1 conta	ains the conference URI an	d the mo	e. SIP#2 receives a REFER request ethod value BYE in the Refer-To conference event package is not	
SIP Parameter	Dial string param	neters options=PIXIT			
	TYPE_SDP= PI>				
	PIXIT for supported header:				
	Case a) No				
		pported: 100 rel;			
		pported: 100 rel and precor	ndition.		
	SIP header valu			WO LID!	
		uest URI contained the UR			
		er-To contains the conferen erred-By contains the URI o			
		nt contains conference; Sub			
		sage/sipfrag contains SIP/2			
		P-Asserted-Identity contain			
		nt contains conference; Sub			
		sage/sipfrag contains SIP/2			
	The	P-Asserted-Identity contain	ns the co	onference URI.	
Comments:		_		OID #G	
SIP#1		Focus		SIP#2	
Conference creation INVITE	→	INVITE			
200 OK (INVITE)	-	200 OK (INVITE)			
ACK (INVITE)	÷	ACK			
SUBSCRIBE	→	AGA.			
200 OK (SUBSCRIBE)					
SIP#2 joining in the cor					
		INVITE	€	INVITE 2	
		180 Ringin		5 5	
		200 OK INVITE			
		ACI	< ←	ACK	
Conference communica					
SIP#1 wishes to remov REFER 1	e SIP#2 from the	conterence	_	DECED	
202 Accepted	7		→ ←	REFER 202 Accepted	
NOTIFY	-		-	NOTIFY 1	
200 OK NOTIFY	÷		÷	200 OK NOTIFY	
	•		-		
			←	BYE	
			→	200 OK BYE	
NOTIFY	←		←	NOTIFY 2	
200 OK NOTIFY	→		→	200 OK NOTIFY	

6.5.2.11.5 Removing a conference participant from a conference

SSSXXSSCONF_ REMOV_001		erence to: ise 5.3.2.6			
TSS reference:		ementary_services/ C0	ONF.		
Configuration	CONF				
Selection criteria:	CONF				
Test purpose:		A participant asks the focus to remove a participant from the conference. The conference event package is subscribed.			
	the conference. In t			to removes an other participant from request to this participant. The	
SIP Parameter	Dial string paramet				
values:	TYPE_SDP= PIXIT	- -;			
	Case c) Supp SIP header values	00 rel; ported: 100 rel; ported: 100 rel and pred		ference URI.	
	Referre NOTIFY 1 Event		l of SIP#1 Subscription	URI. on-State contains active	
		ge/sipfrag contains SII			
				on-State contains active	
	NOTIFY 3 Event	ge/sipfrag contains SII contains conference; cument: <user> uri=S</user>	Subscription	on-State contains terminated	
	NOTIFY 4 Event		Subscription	on-State contains active	
Comments:				•	
SIP#1		Focus		SIP#2	
				~ · · · · · ·	
Conference creation					
INVITE		INVITE			
INVITE 200 OK (INVITE)	← 2	INVITE 200 OK (INVITE)			
INVITE 200 OK (INVITE) ACK	← 2	INVITE			
INVITE 200 OK (INVITE) ACK SUBSCRIBE	← 2 → /	INVITE 200 OK (INVITE)			
INVITE 200 OK (INVITE) ACK	← 2 → / → ←	INVITE 200 OK (INVITE) ACK	foronco		
INVITE 200 OK (INVITE) ACK SUBSCRIBE	← 2 → / → ←	INVITE 200 OK (INVITE) ACK IP#2 joining in the con			
INVITE 200 OK (INVITE) ACK SUBSCRIBE	← 2 → / → ←	INVITE 200 OK (INVITE) ACK IP#2 joining in the con INVI 180 Ringi 200 OK INVI	TE ← ng → TE →	INVITE 2 180 Ringing 200 OK INVITE	
INVITE 200 OK (INVITE) ACK SUBSCRIBE	← 2 → / → ←	INVITE 200 OK (INVITE) ACK IP#2 joining in the con INVI 180 Ringi 200 OK INVI	TE ← ng → TE → CK ←	INVITE 2 180 Ringing	
INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE REFER 1 202 Accepted NOTIFY		INVITE 200 OK (INVITE) ACK SIP#2 joining in the con INVI 180 Ringi 200 OK INVI AC Conference communi es to remove SIP#2 fro REFER 202 Accepted NOTIFY 1	TE ← ng → TE → CK ← caton	INVITE 2 180 Ringing 200 OK INVITE ACK	
INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE REFER 1 202 Accepted	SIP#1 wisher → 1 ← 2 ← 1 → 2	INVITE 200 OK (INVITE) ACK IP#2 joining in the con INVI 180 Ringi 200 OK INVI At Conference communi es to remove SIP#2 fro REFER 202 Accepted	TE ← ng → TE → CK ← caton om the con	INVITE 2 180 Ringing 200 OK INVITE ACK ference	
INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE REFER 1 202 Accepted NOTIFY	SIP#1 wisher + 2 + 2 + 2 + 1 + 2 + 1 + 2 + 1 + 2 + 1 + 2 + 1 + 2 + 1 + 2 + 1 + 2 + 1 + 2 + 3 + 4 + 5 + 6 + 7	INVITE 200 OK (INVITE) ACK IP#2 joining in the con INVI 180 Ringi 200 OK INVI AC Conference communi es to remove SIP#2 fro REFER 202 Accepted NOTIFY 1 200 OK NOTIFY	TE ← ng → TE → CK ← caton om the con	INVITE 2 180 Ringing 200 OK INVITE ACK ference	
INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE REFER 1 202 Accepted NOTIFY 200 OK NOTIFY	SIP#1 wisher + +	INVITE 200 OK (INVITE) ACK IP#2 joining in the con INVI 180 Ringi 200 OK INVI AC Conference communi es to remove SIP#2 fro REFER 202 Accepted NOTIFY 1 200 OK NOTIFY focus removes SIP#2 f	TE ← ng → TE → CK ← caton om the con	INVITE 2 180 Ringing 200 OK INVITE ACK ference	

SSS_XXSSCONF_		reference to:		
REMOV_002 TSS reference:		clause 5.3.2.5 upplementary_services/ C	ONE	
Configuration	CONF	upplementary_services/ C	OINI .	
Selection criteria:	CONF			
Test purpose:		sks the focus to remove a	narticina	ant from the conference. The
1000 pui pood.	conference ever SIP#2 asks the from the confer	ent package is not subscribe focus by sending a REFE rence. In this case the focu	ed. R reque s sends	est to removes an other participant a BYE request to this participant.
SIP Parameter values:		e event package is not sub meters options=PIXIT	Scribed	
	TYPE_SDP= P PIXIT for support Case a) N Case b) S Case c) S SIP header va REFER 1: Rec Ref Ref NOTIFY 1 Eve me: NOTIFY 2 Eve	PIXIT; orted header: lo 100 rel; Supported: 100 rel; Supported: 100 rel and pre lues: quest URI contained the U fer-To contains the SIP#2 ferred-By contains the URI ent contains conference; ssage/sipfrag contains SIF	RI of co URI, me of SIP# Subscrip P/2.0 100 Subscrip	nference URI. thod=BYE. to URI. tion-State contains active Trying. tion-State contains active
Comments:		F		CID#2
SIP#1		Focus Conference creation		SIP#2
INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE)	→ + → +	INVITE 200 OK (INVITE) ACK	ı	
200 OK (OODOOKIDE)	=	SIP#2 joining in the confe	erence	
		INVITI 180 Ringin 200 OK INVITI ACI	g + + + + + + + + + + + + + + + + + + +	INVITE 2 180 Ringing 200 OK INVITE ACK
	015	Conference communic		
DEEED 4		es to remove SIP#2 from	the co	nterence
REFER 1 202 Accepted	→	REFER 202 Accepted		
NOTIFY	-	NOTIFY 1		
200 OK NOTIFY	→	200 OK NOTIFY		
			ves SIP	#2 from the conference BYE
	_		←	200 OK BYE
NOTIFY 200 NOTIFY	← →	NOTIFY 2 200 NOTIFY		

REMOV_003	[20] clau	erence to: use 5.3.2.6		
		ementary_services/ CO	NF.	
Configuration	CONF			
Selection criteria:	CONF			
Test purpose:	The focus removes a participant from the conference. The conference event package is subscribed. The focus wishes to removes an other participant from the conference. In this case the focus sends a BYE request to this participant. The conference event package is subscribed.			
	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.			
	SIP header values:			
	NOTIFY 1 Event contains conference ; Subscription-State contains terminated			
	xml document: <user></user> uri=SIP#2, status is departed .			
	NOTIFY 2 Event contains conference ; Subscription-State contains active			
	xml document: <user></user> uri=SIP#1, status is departed.			
Comments: SIP#1		Focus		SIP#2
Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE)	→ ← → +	INVITE 200 OK (INVITE) ACK		
200 OK (OODOOKIDE)		IP#2 joining in the co	nference	
	Š	IN' 180 Rir 200 OK IN' Conference commun	VITE ← nging → VITE → ACK ← nicaton	INVITE 2 180 Ringing 200 OK INVITE ACK P#2 from the conference
		iocus ii		BYF
			+	200 OK BYE
NOTIFY 200 NOTIFY	← →	NOTI 200 NO NOTIFY 2 200 NOTIFY		NOTIFY 200 NOTIFY

SSS_XXSSCONF_	NGN reference to:		
REMOV_004 TSS reference:	[20] clause 5.3.2.6 SIP-SIP-SIP/Supplementary_services/ (CONE	
	CONF	JONE.	
Configuration Selection criteria:	CONF		
Test purpose:	The focus removes a participant from the	o conforces	The conference event
rest purpose.	package is not subscribed.	e comerence	e. The comerence event
	The focus wishes to removes an other p		
	the focus sends a BYE request to this p not subscribed.	articipant. Th	e conference event package is
SIP Parameter	Dial string parameters options=PIXIT		
values:	TYPE_SDP= PIXIT;		
	PIXIT for supported header:		
	Case a) No 100 rel;		
	Case b) Supported: 100 rel;	1141	
0	Case c) Supported: 100 rel and pr	econdition.	
Comments: SIP#1	Focus		SIP#2
011 #1	Conference creation	1	Oli #Z
INVITE	→ INVITE	•	
200 OK (INVITE)	← 200 OK (INVITE)		
ACK	→ ACK		
SUBSCRIBE	→		
200 OK (SUBSCRIBE)	←		
	SIP#2 joining in the confe	erence	
			INVITE 2
	180 Rir	nging →	180 Ringing
	200 OK IN		200 OK INVITE
			ACK
	Conference communic		
	tocus remo		rom the conference
		→	BYE 200 OK BYE
		~	200 ON DIE

SSS_XXSSCONF_ REMOV_005	_	reference to: lause 5.3.2.6		
TSS reference:	SIP-SIP-SIP/Sup	plementary_services/ CC	NF.	
Configuration	CONF			
Selection criteria:	CONF			
Test purpose:	conference even	t package ist subscribed.		by sending a BYE to the focus. The the focus. The other focus. The entire conference is
		nference event package i	s subscribe	d.
SIP Parameter values:	TYPE_SDP= PIX PIXIT for suppor Case a) No Case b) Su Case c) Su SIP header valu BYE: Req NOTIFY 1 Eve xml NOTIFY 2 Eve	ted header: 100 rel; pported: 100 rel; pported: 100 rel and preces: uest URI = conference Unit contains conference; Sidocument: <user> uri=SI</user>	RI. Subscription P#2, status Subscription	-State contains terminated
Comments: SIP#1		Focus		SIP#2
INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE)	← → →	Conference creat INVITE 200 OK (INVITE) ACK	ion	
BYE 200 OK BYE	SIP#1 → ←		VITE ← nging → VITE → ACK ← nicaton	200 OK ĬNŬITE ACK
		focus rer		t2 from the conference BYE 200 OK BYE
NOTIFY 200 NOTIFY	← →	NOTI 200 NO NOTIFY 2 200 NOTIFY		NOTIFY 200 NOTIFY

Configuration CO Selection criteria: CO Test purpose: The co	ONF ONF he conference own	mentary_services/ CO er releases the entire	NF.				
Selection criteria: Co Test purpose: The co SI re	ONF he conference own	er releases the entire		CONF			
Test purpose: The constant of	he conference own	er releases the entire	CONF				
SI re		er releases the entire	he conference owner releases the entire conference by sending a BYE to the focus. The				
re		ckage is not subscribe	ed.				
					The entire conference is		
ISIP Parameter Di		ence event package is	s not subs	cribed.			
	ial string paramete	s options=PIXIT					
	YPE_SDP= PIXIT;						
PI	PIXIT for supported header:						
	Case a) No 100	•					
	Case b) Suppo						
CI	Case c) Suppo IP header values:	rted: 100 rel and preco	onaition.				
		IDIfaranca LIDI					
Comments:	YE: Request l	JRI = conference URI					
SIP#1		Focus			SIP#2		
311#1		Conference cre	ation		31F#2		
INVITE	→	INVITE	ation				
200 OK (INVITE)	É	200 OK (INVITE)					
ACK	À	ACK					
7.01.	_	71011					
SIP#2 joining in the	e conference						
, ,		IN	VITE	INVITE 2			
		180 Rir	nging 🗦	180 Ringir	ng		
		200 OK IN		200 OK IN			
			ACK ←	ACK			
		Conference comm	unicaton				
	SIP#1	wishes to finish the e	entire cor	ference			
BYE	→	BYE					
200 OK BYE	←	200 OK BYE					
		focus re			e conference		
			7				
			+	200 OK B	ΥE		

SSS_XXSSCONF_ REMOV_007	NGN reference to: [20] clause 5.3.2.7				
TSS reference:	SIP-SIP-SIP/Supplementary_services/	CONF.			
Configuration	CONF				
Selection criteria:	CONF				
Test purpose:	The conference policy dictates to terminate the entire conference by sending a BYE to the locus. The conference event package ist subscribed.				
	request to each participant. The confe	rminate the entire conference by sending a BYE rence event package is subscribed.			
SIP Parameter values:	xml document: <user> uri NOTIFY 2 Event contains conference</user>	ce; Subscription-State contains terminated =SIP#2, status is departed. ce; Subscription-State contains terminated =SIP#1, status is departed.			
SIP#1	Focus	SIP#2			
Conference creation INVITE 200 OK (INVITE) ACK SUBSCRIBE 200 OK (SUBSCRIBE	→ INVITE ← 200 OK (INVITE) → ACK →				
	SIP#2 joining in the	conference			
BYE 200 OK BYE	180	INVITE ← INVITE 2 Ringing → 180 Ringing (INVITE → 200 OK INVITE ACK ← ACK municaton			
	foci	us removes SIP#2 from the conference → BYE ← 200 OK BYE			
NOTIFY 200 NOTIFY		OTIFY 1 → NOTIFY NOTIFY ← 200 NOTIFY			

SSS_XXSSCONF_	NGN reference to:				
REMOV_008	[20] clause 5.3.2.7				
TSS reference:	SIP-SIP/Supplementary_services/ CONF.				
Configuration	CONF				
Selection criteria:	CONF				
Test purpose:	The conference policy dictates to terminate the entire conference by sending a BYE to the focus. The conference event package ist not subscribed.				
	The conference policy decides to termina each participant. The conference event p				
SIP Parameter	Dial string parameters options=PIXIT				
values:	TYPE_SDP= PIXIT; PIXIT for supported header:				
	Case a) No 100 rel;				
	Case b) Supported: 100 rel;				
	Case c) Supported: 100 rel and pre	condition.			
Comments:	_				
SIP#1	Focus			SIP#2	
INVITE 200 OK (INVITE) ACK	Conference cre → INVITE ← 200 OK (INVITE) → ACK	ation			
	SIP#2 joining in the	onferenc	e		
	180	INVITE Ringing INVITE ACK unicaton	← → → ←	INVITE 2 180 Ringing 200 OK INVITE ACK	
BYE 200 OK BYE	← BYE → 200 OK BYE	Tire Com		.	
	focus removes SIP#2 from the conference				
			→	BYE	
			←	200 OK BYE	

Annex A (informative): Bibliography

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History

Document history				
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