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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

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Contents

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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".

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.
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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.

3) Where it is necessary to refer with finer granularity to components of a SIP message, the component concerned is identified by the ABNF rule name used to designate it in the defining RFC (generally 25/RFC 3261 [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.

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

EXAMPLE 6: 200 OK INVITE, 200 OK PRACK.

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

EXAMPLE 7: m=line, a=line.

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

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

3.3 Abbreviations

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

GW	GateWay
I	Inopportune
IUT	Implementation Under Test
MCU	Multipoint Control Unit
MGCF	Media Gateway Control Function
MSI	Manufacturer Specific Information
PDU	Protocol Data Unit
PER	Packed Encoding Rules
PICS	Protocol Implementation Conformance Statement
PIXIT	Protocol Implementation eXtra Information for Testing
PSA	PSase A : Call setup signalling procedures
PSE	PSase E : Call termination signalling procedures
RAS	Registration, Admission and Status
RCF	Register Confirm
REG	REGistration
RRJ	Register Reject
RRQ	Register Request
S	Syntactically invalid
STA	STAtus
TE	Terminal
TP	Test Purpose
TSS	Test Suite Structure
UCF	Unregistration ConFirm
UE	User Equipment
URJ	Unregistration ReJect
URQ	Unregistration ReQuest
V	Valid

4 Test Suite Structure (TSS)

4.1 SIP-SIP

C – Plane / U – Plane			
Basic_Call	Successful		SS___XX___xx
		Codec negotiation	SS___CN___xx
		UPDATE	SS___XX__UP__xx
		Unsuccessful	SS___XX__Uxx
Supplementary Services		OIP/OIR	SS___XXSS_OIPxx
			SS___XXSS_OIRxx
		TIP/TIR	SS___XXSS_TIPxx
		HOLD	SS___XXSS_CHxx
		CFU	SS___XXSS_CFUxx
		CFB	SS___XXSS_CFBxx
		CFNR	SS___XXSS_CFNRxx
		CFNL	SS___XXSS_CHFNLxx
		CONF	SS___XXSS_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__XX

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

5.3 Supplementary Services

CLIP	IS___XXSSCLIP XX

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

6 Test purposes

The registration and application usage procedures in the ATS shall be compliant to RFC 3261 [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]

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 Values			
	UE (MGCF) originating case	**UE (MGCF) terminating 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 netwotk 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 283 003 [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 defined as : TYPE_SDP is performed correctly. Ensure that in the active call state the voice/data transfer on the media 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 1 applies. The call is released from the **called** 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:	A) SDP pre-condition not requested **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 ← ← 183 Session Progress SDP Authorize QoS resource at P-SCCF 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 ← ← 180 Ringing PRACK → → PRACK 200 OK PRACK ← ← 200 OK PRACK 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK BYE ← ← BYE 200 OK BYE → → 200 OK BYE	

SS___XX__02	**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 defined as : TYPE_SDP is performed correctly. Ensure that in the active call state the voice/data transfer on the media 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 1 applies. The call is released from the **calling** 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:	

A) SDP pre-condition not requested

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	←			
			←	183 Session Progress SDP
		Authorize QoS resource at P-SCCF		
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	←		←	180 Ringing
PRACK	→		→	PRACK
200 OK PRACK	←		←	200 OK PRACK
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS___XX__03	**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.
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

SS___XX__03	**NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1]**	
Comments:		
	A) SDP pre-condition not requested **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	

SS___XX__04	**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 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.	
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 → → INVITE 100 Trying ← 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK BYE ← ← BYE 200 OK BYE → → 200 OK BYE	

Table 1: Values for the test purpose SS___XX__01 to SS___XX__04

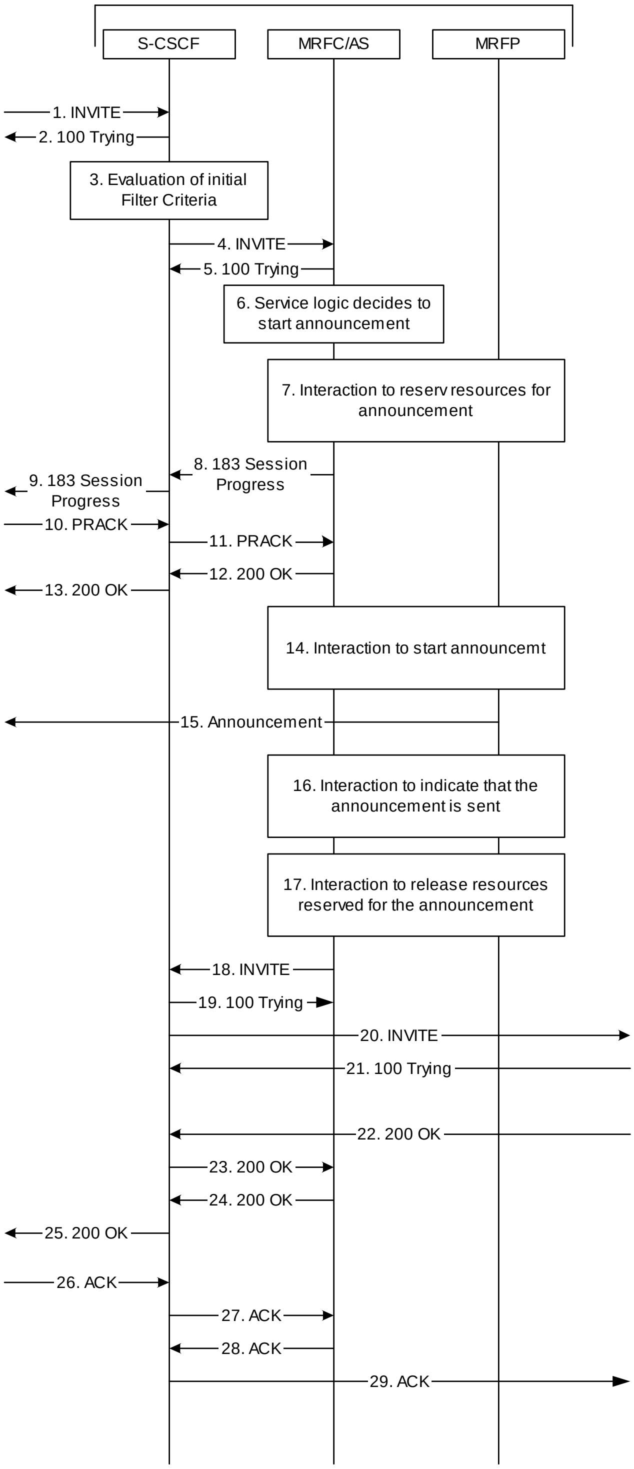
m= line				b= line	a= line
VA	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>
				NOTE: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.	
VA_01	Audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A
VA_02	Audio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

Table 2: Values for test purposes SS___XX__01 and SS___XX__04

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

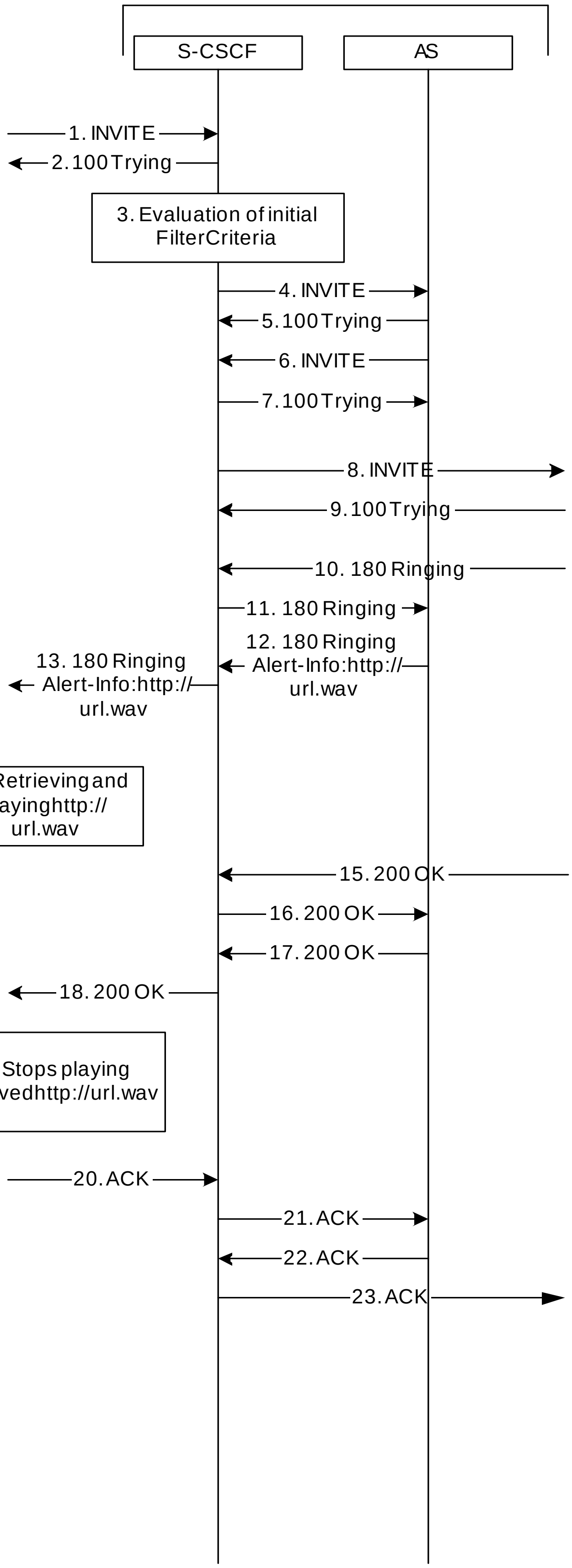
SS___XX__05	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]** **TS 183 028 [18]**	
TSS reference:	SIP-SIP/Basic_call/Successful	
Selection criteria:		
Test purpose:	Ensure that an AS can send an announcement to the calling user during the establishment of a communication. The 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.	

Comments:

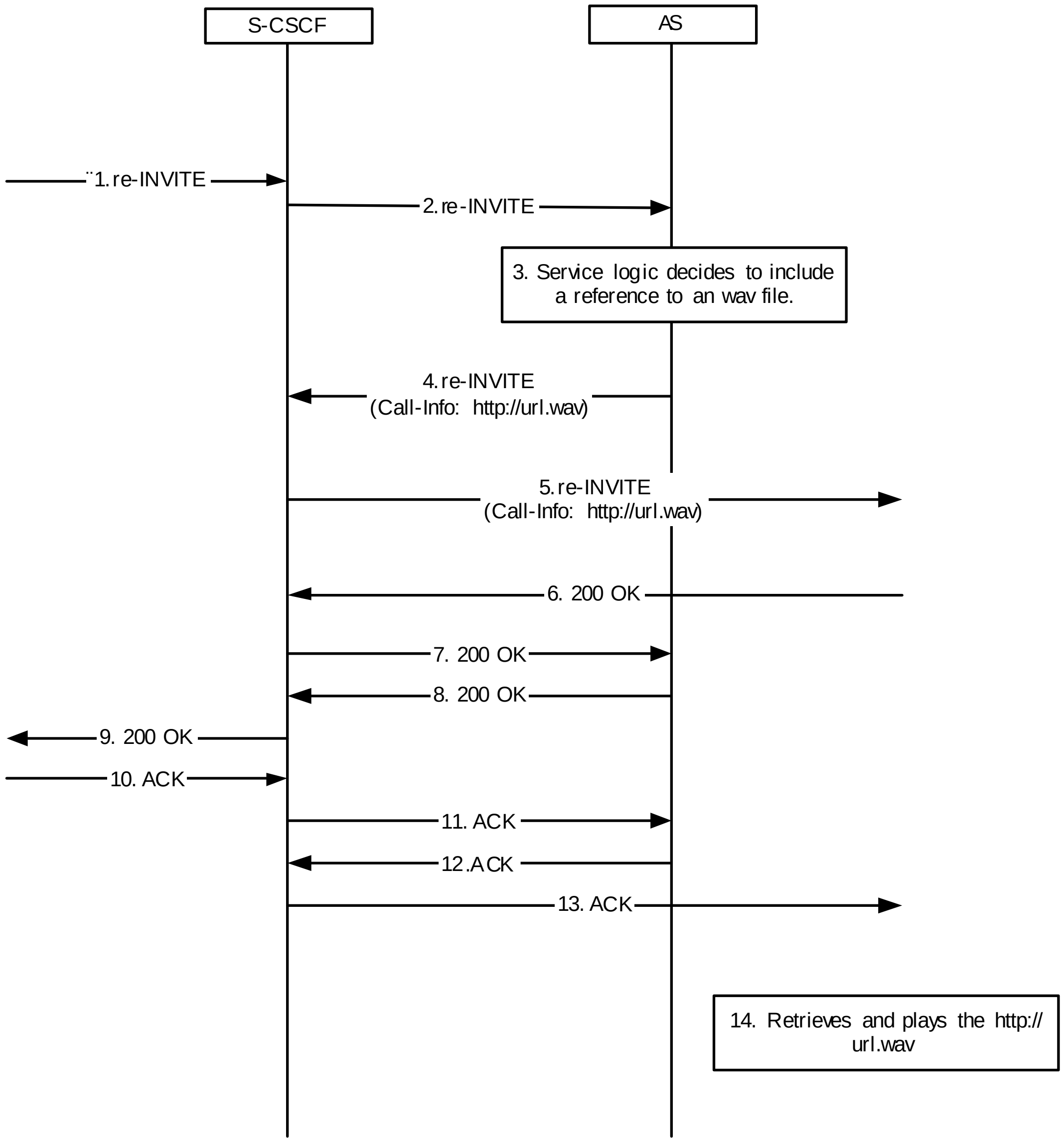

S-CSCF
MRFC/AS
MRFP
1. INVITE
2. 100 Trying
3. Evaluation of initial Filter Criteria
4. INVITE
5. 100 Trying
6. Service logic decides to start announcement
7. Interaction to reserv resources for announcement
8. 183 Session Progress
9. 183 Session Progress
10. PRACK
11. PRACK
12. 200 OK
13. 200 OK
14. Interaction to start announcemt
15. Announcement
16. Interaction to indicate that the announcement is sent
17. Interaction to release resources reserved for the announcement
18. INVITE
19. 100 Trying
20. INVITE
21. 100 Trying
22. 200 OK
23. 200 OK
24. 200 OK
25. 200 OK
26. ACK
27. ACK
28. ACK
29. ACK

SS___XX__06	**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.	

Comments:


S-CSCF
AS
1. INVITE
2. 100 Trying
3. Evaluation of initial FilterCriteria
4. INVITE
5. 100 Trying
6. INVITE
7. 100 Trying
8. INVITE
9. 100 Trying
10. 180 Ringing
11. 180 Ringing
12. 180 Ringing Alert-Info:http:// url.wav
13. 180 Ringing Alert-Info:http:// url.wav
14. Retrieving and playing http:// url.wav
15. 200 OK
16. 200 OK
17. 200 OK
18. 200 OK
19. Stops playing retrieved http://url.wav
20. ACK
21. ACK
22. ACK
23. ACK

SS___XX__07	**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:	Providing announcements to a user during an established communication.	
Test purpose:	Two users are communicating with (at least) one AS in the signalling path. Ensure that a user can place a communication session on hold and sends a re-INVITE request towards the remote user involved in the communication. The *Call-Info header field in a re-INVITE request to indicate media.*	
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:		



6.5.1.1 Codec negotiation

<table>
<tr><td>SS___CN__01</td><td>NGN reference to:
RFC 3261 [2]
TS 124 229 [11] / ES 283 003 [1]
RFC 3311 [24]</td><td></td></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-SIP/Basic_call/Codec negotiation.</td></tr>
<tr><td>Selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2">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 that the other party knowsthat it is to modify an existing session instead of establishing a new session. The other party sends a 200 (OK) to accept the change. The requestor responds to the 200 (OK) with an ACK.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">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.</td></tr>
<tr><td>Comments:</td><td colspan="2">A) SDP pre-condition not requested
<table>
<tr><th>SIP</th><th></th><th>SUT</th><th></th><th>SIP</th></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>re-INVITE</td><td>→</td><td></td><td>→</td><td>re-INVITE</td></tr>
<tr><td>200 OK</td><td>←</td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
</table>
C) pre-condition and 100 rel
<table>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress
SDP</td></tr>
<tr><td></td><td colspan="3">Authorize QoS resource at P-SCCF</td><td></td></tr>
<tr><td>183 Session Progress
SDP</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td colspan="3">Resource Reservation at UE1</td><td></td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td>UPDATE</td><td>→</td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td>200 OK UPDATE</td><td>←</td><td></td><td>←</td><td>200 OK UPDATE</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>re-INVITE</td><td>→</td><td></td><td>→</td><td>re-INVITE</td></tr>
<tr><td>200 OK</td><td>←</td><td></td><td>←</td><td>200 OK</td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
</table></td></tr>
</table>

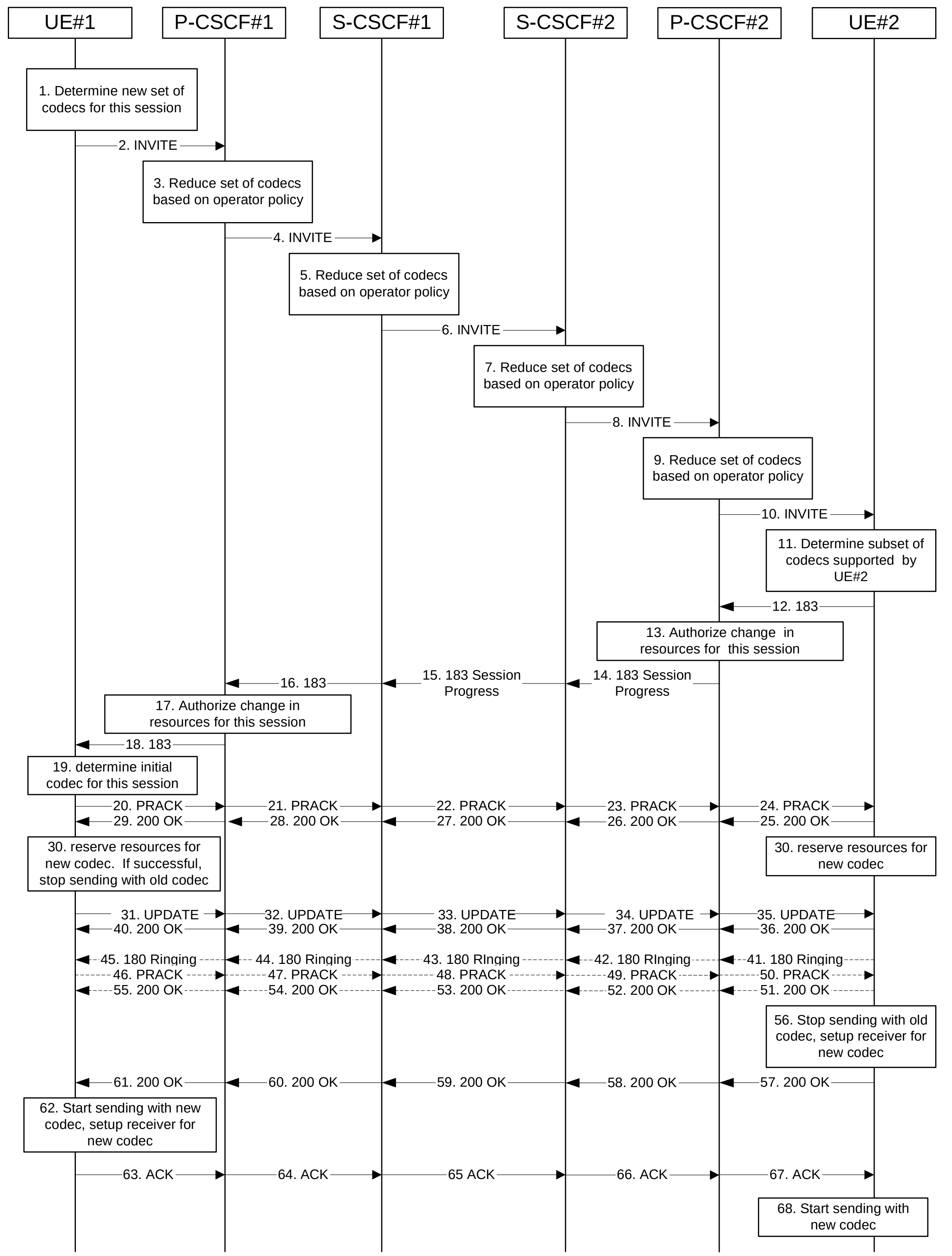
SS___CN__02	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]** **RFC 3311 [24]**	
TSS reference:	SIP-SIP/Basic_call/Codec negotiation	
Selection criteria:		
Test purpose:	During the session, the **called** user decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re- INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. The other party sends a 200 (OK) to accept the change.The requestor responds to the 200 (OK) with an ACK. In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 2 applies.	
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	
	INVITE → → 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 and 100 rel	
	INVITE → → INVITE 100 Trying ← ← 183 Session Progress SDP Authorize QoS resource at P-SCCF 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 ← ← 180 Ringing PRACK → → PRACK 200 OK PRACK ← ← 200 OK PRACK 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK BYE ← ← BYE re-INVITE ← ← re-INVITE 200 OK → → 200 OK 200 OK BYE → → 200 OK BYE	

SS___CN__03	**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 is performed correctly. • The initial INVITE contains a SDP with the offer 1. • The answer related to the **SDP offer is contained in the 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).			
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:	SDP pre-condition not requested			
	SIP	**SUT**		**SIP**
	INVITE offer 1	→	→	INVITE offer1
	100 Trying	←		
	180 Ringing Ringing answer 1	←	←	180 Ringing answer 1
	200 OK INVITE	←	←	200 OK INVITE
	ACK	→	→	ACK
	BYE	←	←	BYE
	A) SDP pre-condition not requested			
	INVITE	→	→	INVITE offer1
	100 Trying	←		
	180 Ringing answer 1	←	←	180 Ringing answer 1
	200 OK INVITE	←	←	200 OK INVITE
	ACK	→	→	ACK
	BYE	←	←	BYE
	200 OK BYE	→	→	200 OK BYE
	C) pre-condition and 100 rel			
	INVITE offer1	→	→	INVITE offer1
	100 Trying	←		
			←	183 Session Progress SDP
	Authorize QoS resource at P-SCCF			
	183 Session Progress SDP	←		
	PRACK	→	→	PRACK
	Resource Reservation at UE1			
	200 OK PRACK	←	←	200 OK PRACK
	180 Ringing answer 1	←	←	180 Ringing answer 1
	PRACK	→	→	PRACK
	200 OK PRACK	←	←	200 OK PRACK
	200 OK INVITE	←	←	200 OK INVITE
	ACK	→	→	ACK
	BYE	←	←	BYE
	200 OK BYE	→	→	200 OK BYE

SS___CN__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 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 contained in the 183 Session Progress** 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 answer 1			←	183 Session Progress answer 1
	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 offer 1	→		→	INVITE offer 1
	100 Trying	←			
				←	183 Session Progress answer 1
			Authorize QoS resource at P-SCCF		
	183 Session Progress SDP answer 1	←			
	PRACK	→		→	PRACK
			Resource Reservation at UE1		
	200 OK PRACK	←		←	200 OK PRACK
	180 Ringing	←		←	180 Ringing
	PRACK	→		→	PRACK
	200 OK PRACK	←		←	200 OK PRACK
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
	BYE	←		←	BYE
	200 OK BYE	→		→	200 OK BYE

SS___CN__05	**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.	

SS___CN__05	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]** **RFC 3311 [24]**	
Comments:		



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TS 124 229 [11] / ES 283 003 [1]
RFC 3311 [24]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-SIP/Basic_call/Codec negotiation.</td></tr>
<tr><td>Selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5">Ensure that the call establishment performed correctly.
• The initial INVITE contains a SDP with the offer 1.
Ensure that answer related to the SDP offer is contained in the 200 OK INVITE 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).</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">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.</td></tr>
<tr><td rowspan="28">Comments:</td><td colspan="5">A) SDP pre-condition not requested</td></tr>
<tr><td>SIP</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>INVITE offer 1</td><td>→</td><td></td><td>→</td><td>INVITE offer 1</td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>200 OK INVITE answer 1</td><td>←</td><td></td><td>←</td><td>200 OK INVITE answer 1</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
<tr><td colspan="5">C) pre-condition and 100 rel</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>100 Trying</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress
SDP</td></tr>
<tr><td colspan="5">Authorize QoS resource at P-SCCF</td></tr>
<tr><td>183 Session Progress
SDP</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>PRACK</td><td>→</td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td colspan="5">Resource Reservation at UE1</td></tr>
<tr><td>200 OK PRACK</td><td>←</td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>200 OK INVITE answer 1</td><td>←</td><td></td><td>←</td><td>200 OK INVITE answer 1</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>BYE</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>200 OK BYE</td></tr>
</table>

SS___CN__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 ← 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 2 200 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 Authorize QoS resource at P-SCCF 183 Session Progress SDP ← PRACK → → PRACK Resource Reservation at UE1 200 OK PRACK ← ← 200 OK PRACK 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 ← ← 200 OK INVITE ACK → → ACK BYE ← ← BYE 200 OK BYE → → 200 OK BYE	

SS___CN__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:		
Test purpose:	Ensure that the call establishment is performed correctly. • Ensure that answer related to the SDP offer is cotained in the 183 Session Progress message. A new offer (codec) is sent in the 180 Ringing. • 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 **answer 1** ← 183 Session Progress **answer 1** 180 Ringing **offer 2** ← ← 180 Ringing **offer 2** 200 OK INVITE ← ← 200 OK INVITE ACK **answer 2** → → ACK **answer 2** BYE ← ← BYE 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** Authorize QoS resource at P-SCCF 183 Session Progress **answer 1** ← PRACK → → PRACK Resource Reservation at UE1 200 OK PRACK ← ← 200 OK PRACK 180 Ringing **offer 2** ← ← 180 Ringing **offer 2** PRACK → → PRACK 200 OK PRACK ← ← 200 OK PRACK 200 OK INVITE ← ← 200 OK INVITE ACK **answer 2** → → ACK **answer 2** BYE ← ← BYE 200 OK BYE → → 200 OK BYE	

6.5.1.2 UPDATE method

SS___XX__UP_01	**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 **contained an offer**. The UPDATE can contain an offer if the callee generated an answer in a reliable provisional response, and the caller has received answers to any other offers it sent in **PRACK** and has generated answers for any offers it received in an UPDATE from 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 precondition.	
Comments:		

SIP		SUT		SIP
INVITE offer 1	→		→	INVITE offer 1
100 Trying	←			
			←	183 Session answer1
		Authorize QoS resource at P-SCCF		
183 Session Progress answer 1	←			
PRACK	→		→	PRACK
200 OK PRACK	←		←	200 OK PRACK
		Resource Reservation at UE1		
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	←		←	200 OK PRACK
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	←		←	BYE
200 OK BYE	→		→	200 OK BYE

SS___XX__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	

SS___XX__UP_02	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]** **RFC 3311 [24]**				
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:** **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/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 UPDATE.				
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition				
Comments:	**SIP**		**SUT**		**SIP**
	INVITE	→		→	INVITE
	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

SS___XX__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 INVITE request for a new dialog, and generates a reliable provisional response containing SDP, that response SHOULD contain an Allow header field that lists the UPDATE method. This informs the caller that the callee is capable of receiving an UPDATE request at any time.	
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 SDP1 with allow header	→		→	INVITE SDP1 with allow header
100 Trying	←			
183 SDP2 with allow header	←		←	183 SDP2 with allow header
PRACK SDP 3	→		→	PRACK SDP 3
200 OK PRACK	←		←	200 OK PRACK
UPDATE SDP 4	→		→	UPDATE SDP 4
200 OK UPDATE	←		←	200 OK UPDATE
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	←		←	BYE

SS___XX__UP_05	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:	SHOULD Tests.	
Test purpose:	Ensure that a 2XX response SHOULD contain an Allow header field listing the UPDATE method.	
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 SDP1 with allow header	→		→	INVITE SDP1 with allow header
100 Trying	←			
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	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
BYE	←		←	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

SS___XX_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 unallocated number, the network initiate call clearing to the calling user with a 404 Not Found 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.				
Comments:	**SIP**		**SUT**		**SIP**
	INVITE	→		→	INVITE
	100 Trying	←			
	404 Not Found	←		←	404 Not Found
	ACK	→		→	ACK

SS___XX_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 Service unavailable. 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.				
Comments:	**SIP**		**SUT**		**SIP**
	INVITE	→		→	INVITE
	100 Trying	←			
	503 Sevice Unawailable	←		←	503 Sevice Unawailable
	ACK	→		→	ACK

SS___XX_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 is busy the **network initiate** call clearing to the calling user with a 486 Busy Here 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.				
Comments:	**SIP**		**SUT**		**SIP**
	INVITE	→		→	INVITE
	100 Trying	←			
	486 Busy Here	←		←	486 Busy Here
	ACK	→		→	ACK

SS___XX_U04	**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 from the called user (there is no response from INVITE messages), the network initiate call clearing to the calling user with a 480 Temporarily unavailable message or 408 request time out.	
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:		

	UE1		**Proxy 1**		**Proxy 2**			**UE2**
INVITE	→							
100	←	INVITE	→					
		100	←	INVITE	→			
						INVITE	→	
						INVITE	→	
						INVITE	→	
						INVITE	→	
						INVITE	→	
						INVITE	→	
						INVITE	→	
480	←	480	←	480	←			
ACK	→	ACK	→	ACK	→			

SS___XX_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**		**SUT**		**SIP UA B**
	INVITE	→		→	INVITE
	180 Ringing	←		←	180 Ringing
	480 Teporary unavaible	←		←	480 Teporary unavaible
	ACK	→		→	ACK

SS___XX_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.	

SS___XX_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

SS___XX_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 out of order, the network initiate call clearing to the calling user with a 502 Bad Gateway 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.				
Comments:	**SIP**		**SUT**		**SIP**
	INVITE	→		→	INVITE
	100 Trying	←			
	502 Bad Gateway	←		←	502 Bad Gateway
	ACK	→		→	ACK

SS___XX_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 when the called number is incomplete. The network initiates call clearing to the calling user with 484 Address Incomplete 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.				
Comments:	**SIP**		**SUT**		**SIP**
	INVITE	→		→	INVITE
	100 Trying	←			
	484 Address Inconplete	←		←	484 Address Inconplete
	ACK	→		→	ACK

SS___XX_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.

SS___XX_U09	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]**	
Comments:	see flow below	

SIP		**SUT**		**SIP**
INVITE	→		→	INVITE
100 Trying	←			
503 Service unavailable	←		←	503 Service unavailable
ACK	→		→	ACK

SS___XX_U10	**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 calling user clears the call with a CANCEL message before answer from called user, the network transport the CANCEL message to the called 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:	Case a)

	UE1		**Proxy 1**		**Proxy 2**		**UE2**
INVITE	→						
100	←	INVITE	→				
		100	←	INVITE	→		
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
CANCEL	→	CANCEL	→	CANCEL	→	CANCEL	→
				Case b)			
INVITE	→						
100	←	INVITE	→				
		100	←	INVITE	→		
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
CANCEL	→	CANCEL	→	CANCEL	→	180	←
						CANCEL	→
				Case c)			
INVITE	→						
100	←	INVITE	→				
		100	←	INVITE	→		
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
CANCEL	→	CANCEL	→	CANCEL	→	200 OK	←
						CANCEL	→

SS___XX_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 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		**SUP**		**SIP**
INVITE	→		→	INVITE
100 Trying	←			
183 Session Progress	←		←	183 Session Progress
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
		Communication		
RE-INVITE offer 2	→		→	RE-INVITE offer 2
488 Not Acceptable Here	←		←	488 Not Acceptable Here
		Communication		
BYE	←		←	BYE
200 OK BYE				200 OK BYE

SS___XX_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	→		→	INVITE
100 Trying	←			
183 Session Progress	←		←	183 Session Progress
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
		Communication		
RE-INVITE offer 2	←		←	RE-INVITE offer 2
488 Not Acceptable Here	→		→	488 Not Acceptable Here
		Communication		
BYE	←		←	BYE
200 OK BYE				200 OK BYE

SS___XX_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 related to the SDP offer is cotained in the 180 Ringing message. The media stream is rejected (port number is set to zero). • Ensure that the call is rejected by sending a CANCEL or BYE.	
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 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	

SS___XX_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 to the SDP offer is cotained in the 183 Session Progress message. The media stream is rejected (port number is set to zero). • Ensure that the call is rejected by sending a CANCEL or BYE.	
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 offer1 → → INVITE offer1 100 Trying ← 183 Session Progress answer 1 ← ← 183 Session Progress 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	

SS___XX_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 to the SDP offer is cotained in the 200 OK INVITE message. The media stream is rejected (port number is set to zero). • Ensure that the call is rejected by sending a BYE.	
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 offer1	→		→	INVITE offer1
100 Trying	←			
200 OK answer 1	→		→	200 OK answer 1
ACK	→		→	ACK
BYE	←		←	BYE
200 OK	→		→	200 OK

SS___XX_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 from the called user ("*no user responding*"), the calling user initiate call clearing to the calling user with CANCEL or BYE.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition.	
Comments:		

SIP		SUT		SIP
INVITE	→		→	INVITE
100 Trying	←			
180 Ringing	←		←	180 Ringing
CANCEL	→		→	CANCEL
200 OK CANCEL	→		→	200 OK CANCEL
487 Request Terminated	←		←	487 Request Terminated
ACK	→		→	ACK

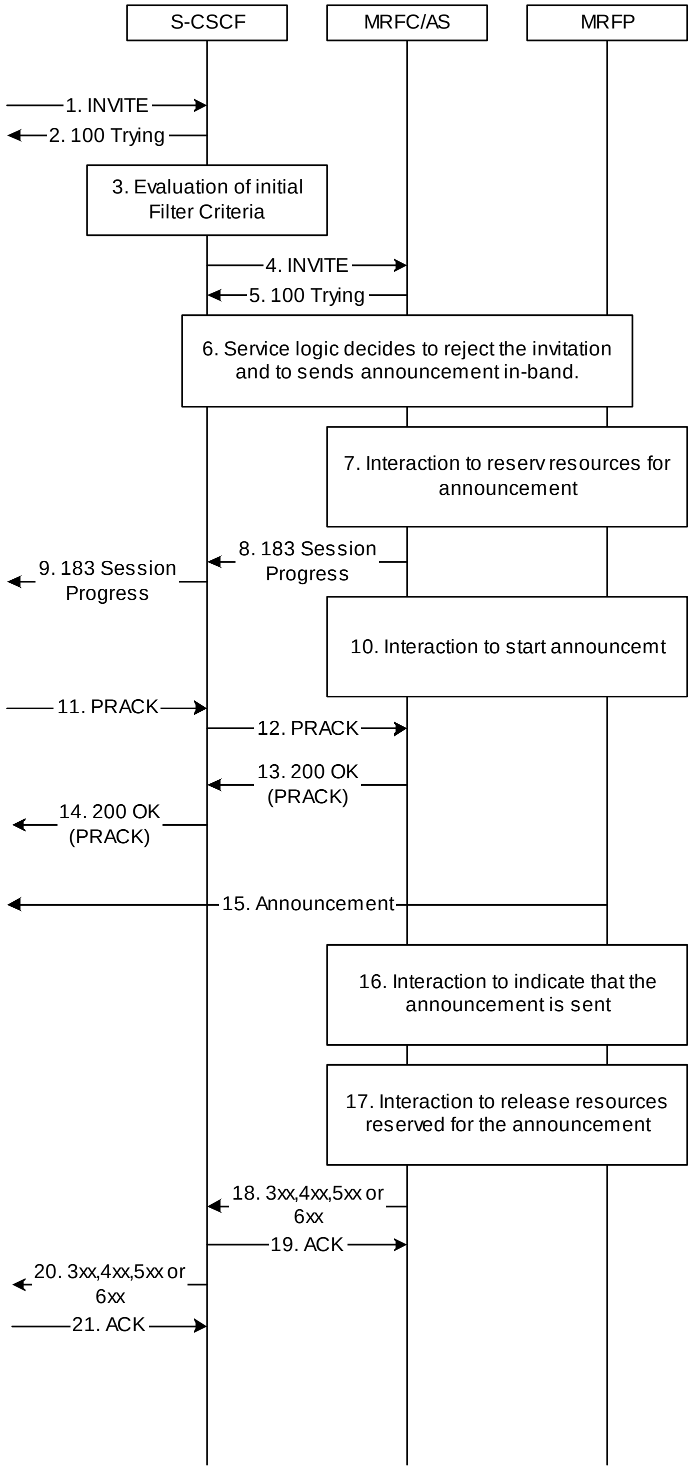
SS___XX_U17	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]**	
TSS reference:	SIP-SIP/Basic_call/Unsuccessful	
Selection criteria:		
Test purpose:	Ensure that upon receiving a 421 (Extension Required) response to an initial INVITE request in which the precondition mechanism 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:		

SI___XX_U18	**PSTN reference to: EN 300 403-1 [14], clause G.1.6**	**NGN reference to: EN 383 001 [15], clause 6.11 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 **calling user** sends a media type not supporte bye the SUT. The call attempt is rejected bye sending a 415 Unsupported media Type.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition..	
Comments:	**SIP** **SUT** **SIP** INVITE → → INVITE 100 Trying ← 415 Unsupported Media Type ← ← 415 Unsupported Media Type ACK → → ACK	

SS___XX_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.	

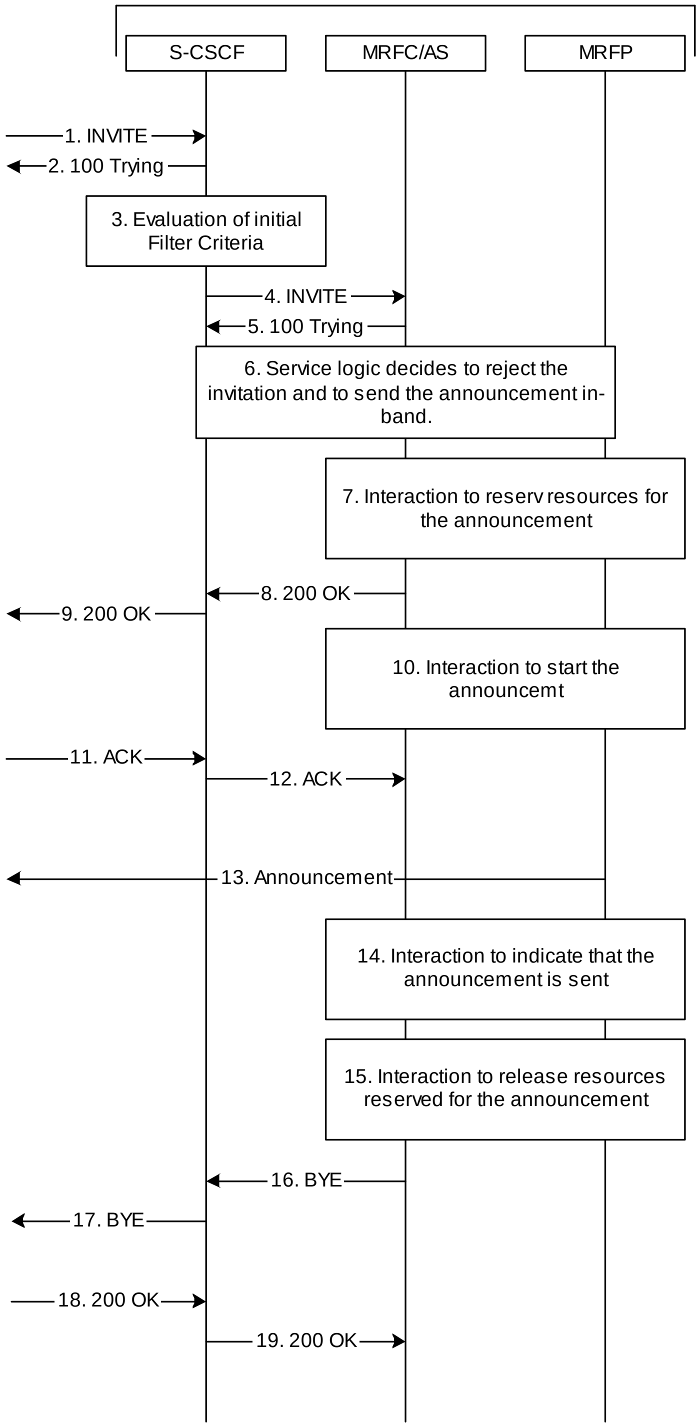
SS___XX_U19	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1] TS 183 028 [18]	

Comments:

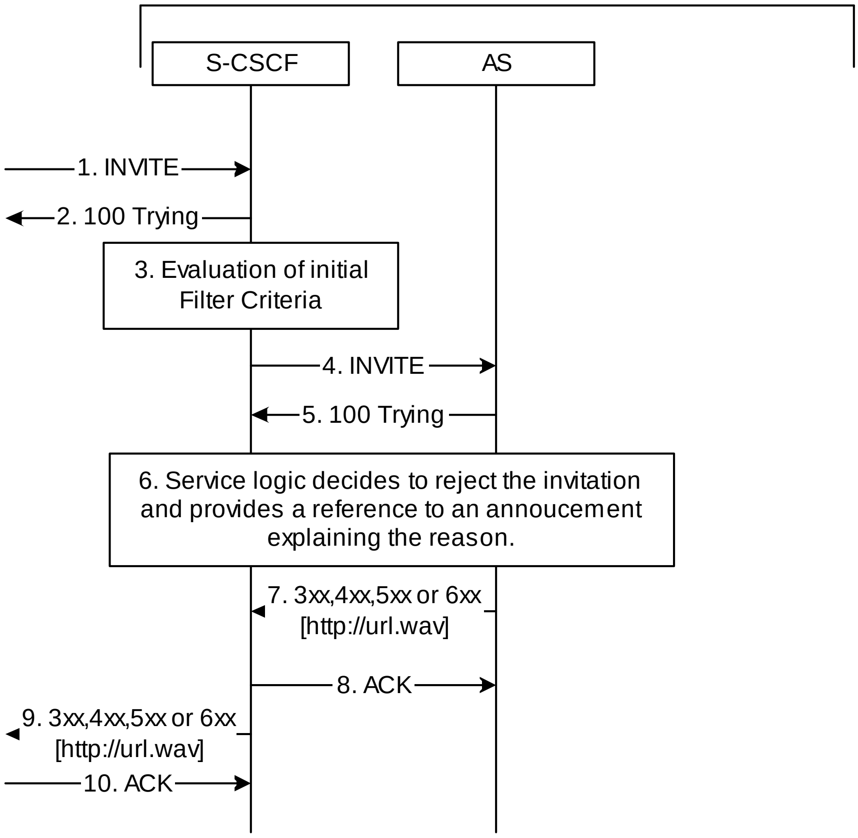


SS___XX_U20	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]** **TS 183 028 [18]**	
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.	
Selection criteria:	• Communication request rejected. • Sending the announcement as in-band information.	
Test purpose:	Service logic in an AS, 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.	

SS___XX_U20	NGN reference to: RFC 3261 [2] TS 124 229 [11] / ES 283 003 [1] TS 183 028 [18]	
Comments:		



SS___XX_U21	**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 include a reference to the announcement in a 3XX, 4XX, 5XX and 6XX 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:		



SS___XX_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 unallocated number, the network initiate call clearing to the calling user with a 404 Not Found 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.			
Comments:	**SIP**		**SUT**	**SIP**
	INVITE	→	→	INVITE
	100 Trying	←		
	404 Not Found	←	←	404 Not Found
	ACK	→	→	ACK

SS___XX_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 Service unavailable. 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.			
Comments:	**SIP**		**SUT**	**SIP**
	INVITE	→	→	INVITE
	100 Trying	←		
	503 Sevice Unawailable	←	←	503 Sevice Unawailable
	ACK	→	→	ACK

SS___XX_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 is busy the **network initiate** call clearing to the calling user with a 486 Busy Here 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.			
Comments:	**SIP**		**SUT**	**SIP**
	INVITE	→	→	INVITE
	100 Trying	←		
	486 Busy Here	←	←	486 Busy Here
	ACK	→	→	ACK

SS___XX_U04	**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 from the called user (there is no response from INVITE messages), the network initiate call clearing to the calling user with a 480 Temporarily unavailable message or 408 request time out .	
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:		

	UE1		**Proxy 1**		**Proxy 2**		**UE2**
INVITE	→						
100	←	INVITE	→				
		100	←	INVITE	→		
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
480	←	480	←	480	←		
ACK	→	ACK	→	ACK	→		

SS___XX_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 SUT SIP UA B INVITE → → INVITE 180 Ringing ← ← 180 Ringing 480 Teporary unavaible ← ← 480 Teporary unavaible ACK → → ACK	

SS___XX_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.	

SS___XX_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

SS___XX_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 out of order, the network initiate call clearing to the calling user with a 502 Bad Gateway 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.				
Comments:	**SIP**		**SUT**		**SIP**
	INVITE	→		→	INVITE
	100 Trying	←			
	502 Bad Gateway	←		←	502 Bad Gateway
	ACK	→		→	ACK

SS___XX_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 when the called number is incomplete. The network initiates call clearing to the calling user with 484 Address Incomplete 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.				
Comments:	**SIP**		**SUT**		**SIP**
	INVITE	→		→	INVITE
	100 Trying	←			
	484 Address Inconplete	←		←	484 Address Inconplete
	ACK	→		→	ACK

SS___XX_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.

SS___XX_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

SS___XX_U10	**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 calling user clears the call with a CANCEL message before answer from called user, the network transport the CANCEL message to the called 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:		

Case a)

	UE1		**Proxy 1**		**Proxy 2**		**UE2**
INVITE	→						
100	←	INVITE	→				
		100	←	INVITE	→		
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
CANCEL	→	CANCEL	→	CANCEL	→	CANCEL	→

Case b)

INVITE	→						
100	←	INVITE	→				
		100	←	INVITE	→		
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
CANCEL	→	CANCEL	→	CANCEL	→	180	←
						CANCEL	→

Case c)

INVITE	→						
100	←	INVITE	→				
		100	←	INVITE	→		
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
						INVITE	→
CANCEL	→	CANCEL	→	CANCEL	→	200 OK	←
						CANCEL	→

SS___XX_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 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:	see flow below	

SIP		**SUP**		**SIP**
INVITE	→		→	INVITE
100 Trying	←			
183 Session Progress	←		←	183 Session Progress
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
		Communication		
RE-INVITE offer 2	→		→	RE-INVITE offer 2
488 Not Acceptable Here	←		←	488 Not Acceptable Here
		Communication		
BYE	←		←	BYE
200 OK BYE				200 OK BYE

SS___XX_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:	see flow below	

SIP		**SUT**		**SIP**
INVITE	→		→	INVITE
100 Trying	←			
183 Session Progress	←		←	183 Session Progress
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
ACK	→		→	ACK
		Communication		
RE-INVITE offer 2	←		←	RE-INVITE offer 2
488 Not Acceptable Here	→		→	488 Not Acceptable Here
		Communication		
BYE	←		←	BYE
200 OK BYE				200 OK BYE

SS___XX_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 related to the SDP offer is cotained in the 180 Ringing message. The media stream is rejected (port number is set to zero). • Ensure that the call is rejected by sending a CANCEL or BYE.				
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 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

SS___XX_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 to the SDP offer is cotained in the 183 Session Progress message. The media stream is rejected (port number is set to zero). • Ensure that the call is rejected by sending a CANCEL or BYE.				
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 offer1	→		→	INVITE offer1
	100 Trying	←			
	183 Session Progress answer 1	←		←	183 Session Progress 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

SS___XX_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 to the SDP offer is cotained in the 200 OK INVITE message. The media stream is rejected (port number is set to zero). • Ensure that the call is rejected by sending a BYE.	
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 offer1 → → INVITE offer1 100 Trying ← 200 OK answer 1 → → 200 OK answer 1 ACK → → ACK BYE ← ← BYE 200 OK → → 200 OK	

SS___XX_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 from the called user ("*no user responding*"), the calling user initiate call clearing to the calling user with CANCEL or BYE.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition.	
Comments:	**SIP** **SUT** **SIP** INVITE → → INVITE 100 Trying ← 180 Ringing ← ← 180 Ringing CANCEL → → CANCEL 200 OK CANCEL → → 200 OK CANCEL 487 Request Terminated ← ← 487 Request Terminated ACK → → ACK	

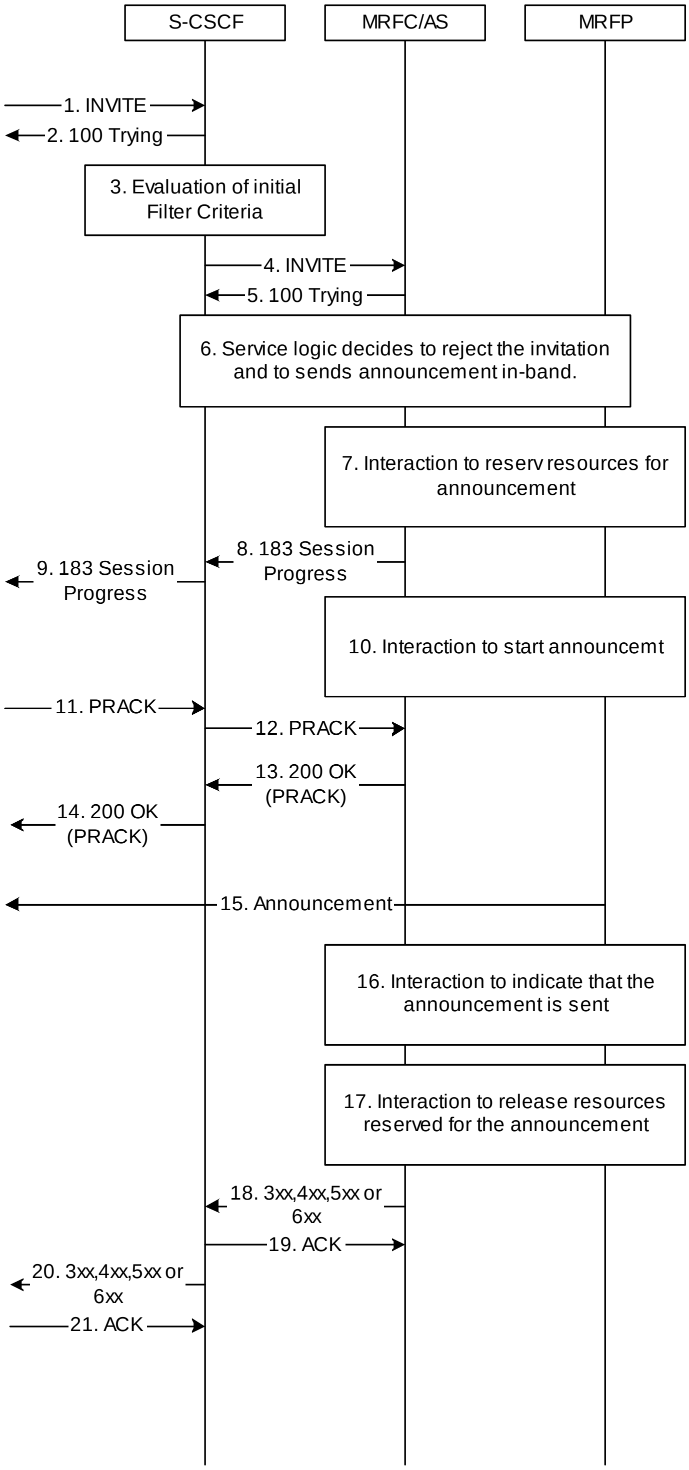
SS___XX_U17	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 / ES 283 003 [1]**	
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.	
Selection criteria:		
Test purpose:	Ensure that upon receiving a 421 (Extension Required) response to an initial INVITE request in which the precondition mechanism was 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:		

SI___XX_U18	**PSTN reference to:** **EN 300 403-1 [14],** **clause G.1.6**	**NGN reference to:** **Q.1912.5 clause 6.11** **EN 383 001 [15] clause 6.11** **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 **calling user** sends a media type not supporte bye the SUT. The call attempt is rejected bye sending a 415 Unsupported media Type.	
SIP Parameter values:	Dial string parameters options=PIXIT TYPE_SDP= PIXIT; PIXIT for supported header: Case a) No 100 rel; Case b) Supported: 100 rel; Case c) Supported: 100 rel and precondition.	
Comments:	**SIP** **SUT** **SIP** INVITE → → INVITE 100 Trying ← 415 Unsupported Media Type ← ← 415 Unsupported Media Type ACK → → ACK	

SS___XX_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.	

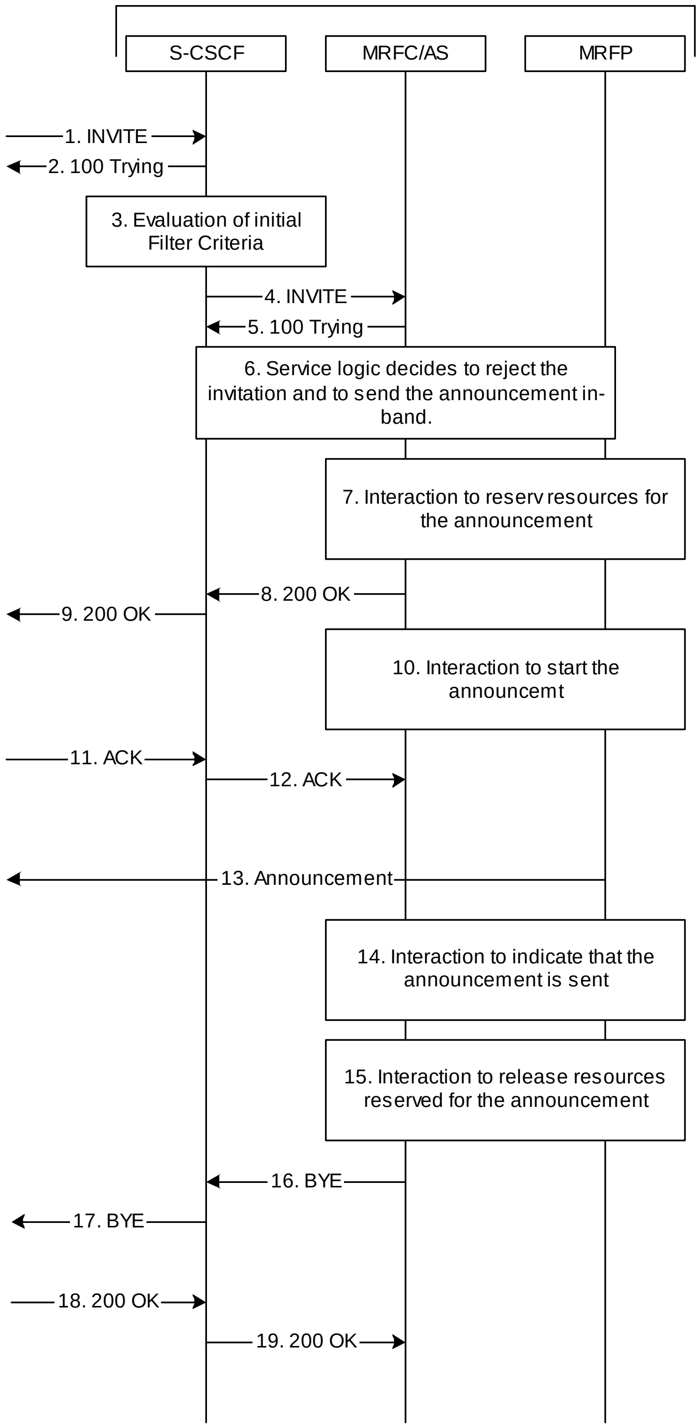
SS___XX_U19	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]** **TS 183 028 [18]**	

Comments:

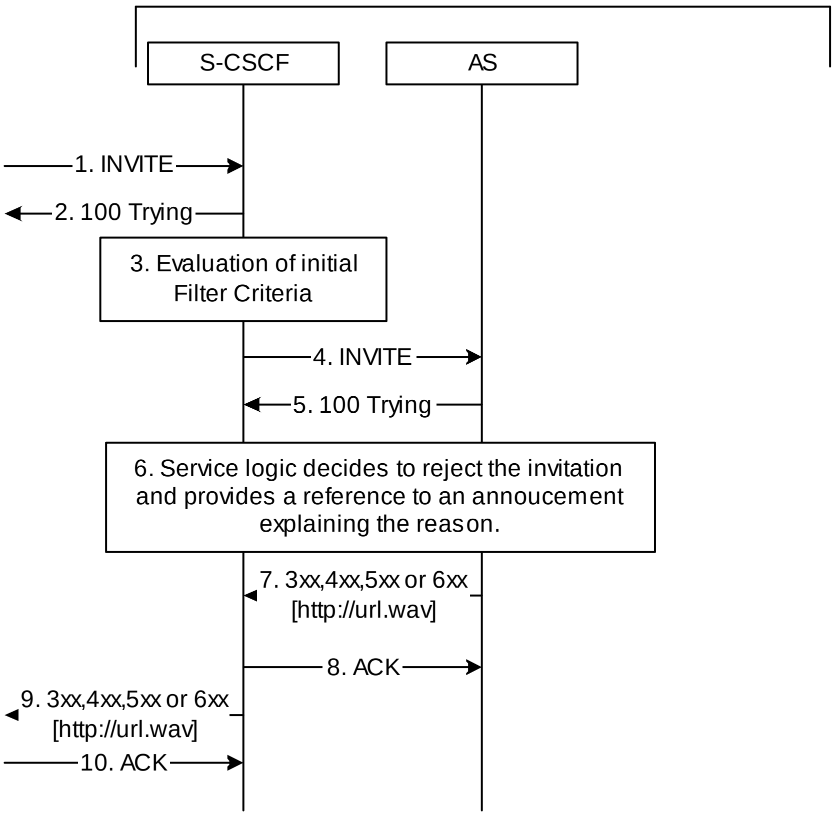


SS___XX_U20	**NGN reference to:** **RFC 3261 [2]** **TS 124 229 [11] / ES 283 003 [1]** **TS 183 028 [18]**	
TSS reference:	SIP-SIP/Basic_call/Unsuccessful.	
Selection criteria:	• Communication request rejected. • Sending the announcement as in-band information.	
Test purpose:	Service logic in an AS, 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.	

Comments:


S-CSCF
MRFC/AS
MRFP
1. INVITE
2. 100 Trying
3. Evaluation of initial Filter Criteria
4. INVITE
5. 100 Trying
6. Service logic decides to reject the invitation and to send the announcement in-band.
7. Interaction to reserv resources for the announcement
8. 200 OK
9. 200 OK
10. Interaction to start the announcemt
11. ACK
12. ACK
13. Announcement
14. Interaction to indicate that the announcement is sent
15. Interaction to release resources reserved for the announcement
16. BYE
17. BYE
18. 200 OK
19. 200 OK

SS___XX_U21	**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 include a reference to the announcement in a 3XX, 4XX, 5XX and 6XX 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:		



6.5.2 Test purposes for SIP-SIP, Supplementary services

6.5.2.1 Test purposes for OIP

SS___XXSS_OIP01	**OIP/OIR reference [16] clauses 4.3.2 and 4.5.2.4**			
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 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 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)

SS___XXSS_OIP02	**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 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 no identity information is provided by the originating UE, the terminating user receives a **P-Asserted-Identity** based on the default public user identity associated with the originating UE 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)

SS___XXSS_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 "temporary mode" default "restricted". 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 no identity information is provided by the originating UE, the terminating user receives a **P-Asserted-Identity** based on the default public user identity associated with the originating UE 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)

SS___XXSS_OIP05	**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 no privacy value is 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 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)

SS___XXSS_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)

SS___XXSS_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 mode" default "restricted". 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=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)

SS___XXSS_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 "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 no privacy value is 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 receives a **P-Asserted-Identity** based on the information provided by 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)

SS___XXSS_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 "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, matches with the set of registered public identities of the originating UE the terminating user receives a **P-Asserted-Identity** based on the information provided by 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)

SS___XXSS_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 "temporary mode" default "restricted". 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, matches with the set of registered public identities of the originating UE the terminating user receives a **P-Asserted-Identity** based on the information provided by 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)

SS___XXSS_OIP13	**OIP/OIR reference [19] clause 4.5.2.3**
TSS reference:	SIP-SIP/SupplementaryServices/OIP.
Selection criteria:	The S-CSCF has knowledge of an associated tel-URI for a SIP URI The terminating user subscribes the OIP service.
Test purpose:	The S-CSCF has knowledge of an associated tel-URI for a SIP URI contained in the P-Asserted-Identity header field received in the request, the S-CSCF adds a second P-Asserted-Identity header field containing this tel-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.
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)

SS___XXSS_OIP14	**OIP/OIR reference [19] clause 4.5.2.4**	
TSS reference:	SIP-SIP/SupplementaryServices/OIP	
Selection criteria:	OIP service subscribed. 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 originating user.	
Test purpose:	The 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 originating user There is no match found, the AS sets the From header to the SIP URI that includes the registered default public user identity.	
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)	

SS___XXSS_OIP15	**OIP/OIR reference [19] clause 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** INVITE → → INVITE 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK Conversation BYE → → BYE 200 OK (BYE) ← ← 200 OK (BYE)	

6.5.2.2 Test purposes for OIR

SS___XXSS_OIR01	**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 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** INVITE → → INVITE 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK Conversation BYE → → BYE 200 OK (BYE) ← ← 200 OK (BYE)

SS___XXSS_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 → → INVITE 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK Conversation BYE → → BYE 200 OK (BYE) ← ← 200 OK (BYE)

SS___XXSS_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 "not 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**		**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)

SS___XXSS_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	→		→ BYE
	200 OK (BYE)	←		← 200 OK (BYE)

SS___XXSS_OIR05	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 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 → → INVITE 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK Conversation BYE → → BYE 200 OK (BYE) ← ← 200 OK (BYE)

SS___XXSS_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 "temporary mode" default "restricted".
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** / **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)

SS___XXSS_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 "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 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 → → INVITE 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK Conversation BYE → → BYE 200 OK (BYE) ← ← 200 OK (BYE)

SS___XXSS_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 "temporary mode" default "not 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** — **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)

SS___XXSS_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 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)

SS___XXSS_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 permanent 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=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)

SS___XXSS_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 "temporary mode" default "restricted".				
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=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)

SS___XXSS_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 "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 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)

SS___XXSS_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 "not 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 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)

SS___XXSS_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 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, 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)

SS___XXSS_OIR15	**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 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**		**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)

SS___XXSS_OIR16	**OIP/OIR reference [19] clause 4.5.2.4**			
TSS reference:	SIP-SIP/SupplementaryServices/OIR.			
Selection criteria:	The user subscribes OIR permanent mode. Modifies the From header field to remove the identification.			
Test purpose:	In case of the OIR service, the AS modifies the From header field to remove the identification information. 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: 10;0 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)

SS___XXSS_OIR17	**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 the AS add a Privacy header field set to "**user**".				
Test purpose:	In case of the OIR service, the AS add a Privacy header 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**		**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)

SS___XXSS_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)

SS___XXSS_OIR19	**OIP/OIR reference [19] clauses 4.3.2 and4.5.2.4**			
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, 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	→		→ INVITE
	180 Ringing	←		← 180 Ringing
	200 OK INVITE	←		← 200 OK INVITE
	ACK	→		→ ACK
			Conversation	
	BYE	→		→ BYE
	200 OK (BYE)	←		← 200 OK (BYE)

SS___XXSS_OIR20	**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 no 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	→		→ INVITE
	180 Ringing	←		← 180 Ringing
	200 OK INVITE	←		← 200 OK INVITE
	ACK	→		→ ACK
			Conversation	
	BYE	→		→ BYE
	200 OK (BYE)	←		← 200 OK (BYE)

6.5.2.3 Test purposes for TIP

SS___XXSS_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 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 precondition.			
Comments:	**SIP UA A**		**SUT**	**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			
	180 Ringing	←		← 180 Ringing
	200 OK INVITE (P-Asserted-Identiy)	←		← 200 OK INVITE
	ACK	→		→ ACK
	Conversation			
	BYE	→		→ BYE
	200 OK (BYE)	←		← 200 OK (BYE)

SS___XXSS_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 service.			
Test purpose:	In case "temporary mode" with default "not restricted" is subscribed and a Privacy header field is set to "**none**", 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 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 precondition.			
Comments:	**SIP UA A**	**SUT**		**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			
	180 Ringing	←	←	180 Ringing
	200 OK INVITE(P-Asserted-Identiy)	←	←	200 OK INVITE
	ACK	→	→	ACK
	Conversation			
	BYE	→	→	BYE
	200 OK (BYE)	←	←	200 OK (BYE)

SS___XXSS_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 service.				
Test purpose:	In case "temporary mode" with default "restricted" is subscribed and a Privacy header field is set to "**none**" 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 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 precondition.				
Comments:	**SIP UA A**		**SUT**		**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				
	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

SS___XXSS_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 service.				
Test purpose:	In case "temporary mode" with default "not restricted" is subscribed and a Privacy header field is set to "**id**", 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	→		→	INVITE
	Case A				
	180 Ringing	←		←	180 Ringing
	200 OK INVITE	←		←	200 OK INVITE
	ACK	→		→	ACK
	Case B				
	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)

SS___XXSS_TIR02	**[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 "restricted". The originating user subscribes the TIP service.
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 → → INVITE **Case A** 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK **Case B** 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)

SS___XXSS_TIR03	**[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 "restricted". The originating user subscribes the TIP service.
Test purpose:	In case "temporary mode" with default "restricted" is subscribed and a Privacy header field is set to "**id**", 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 → → INVITE **Case A** 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK **Case B** 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)

SS___XXSS_TIR04	[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 service.
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 → → INVITE **Case A** 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK **Case B** 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)

SS___XXSS_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 service.
Test purpose:	In case "permanent mode" is subscribed and a Privacy header field is set to "**id**", 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 → → INVITE **Case A** 180 Ringing ← ← 180 Ringing 200 OK INVITE ← ← 200 OK INVITE ACK → → ACK **Case B** 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)

SS___XXSS_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 service. The originating use has the "override category". The terminating user subscribes TIR "permanent mode".			
Test purpose:	Ensure that an originating user, that subscribed the TIP service and additionally has the override category, receives the P-Asserted-Identity identifying the callee in a 1XX response or in a 2XX response. The originating user does not receive a Privacy value "id".			
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
	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			
	180 Ringing	←	←	180 Ringing
	200 OK INVITE(P-Asserted-Identiy)	←	←	200 OK INVITE
	ACK	→	→	ACK
	Conversation			
	BYE	→	→	BYE
	200 OK (BYE)	←	←	200 OK (BYE)

<table>
<tr><td>SS___XXSS_TIR17</td><td colspan="4">[18] clause 5.2.6.4, [12] clause 4.5.2.4</td></tr>
<tr><td>TSS reference:</td><td colspan="4">SIP-SIP/SupplementaryServices/TIP</td></tr>
<tr><td>Selection criteria:</td><td colspan="4">The user subscribes TIR "permanent mode".
The originating user subscribes the TIP service.</td></tr>
<tr><td>Test purpose:</td><td colspan="4">In case "permanent mode" is subscribed and a Privacy header field is set to "none", 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 associated with the terminating UE identifies the callee.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="4">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.</td></tr>
<tr><td>Comments:</td><td>SIP UA A
INVITE
Case A
180 Ringing(P-Asserted-Identiy)
200 OK INVITE
ACK
Case B
183 Session Progress(P-Asserted-Identiy)
180 Ringing
200 OK INVITE
ACK
Case C
180 Ringing
200 OK INVITE(P-Asserted-Identiy)
ACK
Conversation
BYE
200 OK (BYE)</td><td>
→

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←</td><td>SUT</td><td>
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→

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←
→

→
←</td><td>SIP UA B
INVITE

180 Ringing
200 OK INVITE
ACK

183 Session Progress
180 Ringing
200 OK INVITE
ACK

180 Ringing
200 OK INVITE
ACK

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

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 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 *receives* 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=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
UPDATE(**sendonly**)	→		→	UPDATE(sendonly)
200 OK UPDATE (**recvonly**)	←		←	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.* T*he 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 *receives* 200 OK final response containing the SDP with the attribute "a=" inactive.	
Precondition:	• A session was established between user A and user B according to the "basic Call" procedures. • The media stream was previously set to "recvonly". • 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**)	→		→	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__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*. 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 *receives* 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 between user A and user B according to the "basic Call" procedures. • The media stream was previously set to "sendonly". • 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**)	→		→	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	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH04	**NGN reference to: [21]**	
TSS reference:	ServedUser/WithoutAnnounc/WithUPDATE	
Selection criteria:	*Session hold. UPDATE method is used. Individual 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 *receives* 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=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
UPDATE(sendonly)	←		←	UPDATE(sendonly)
200 OK UPDATE (recvonly)	→		→	200 OK UPDATE (recvonly)
UPDATE(**inactive**)	→		→	UPDATE(inactive)
200 OK UPDATE (**inactive**)	←		←	200 OK UPDATE (inactive)
UPDATE(**recvonly**)	→		→	UPDATE(recvonly)
200 OK UPDATE (**sendonly**)	←		←	200 OK UPDATE (sendonly)
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH05	**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 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 *receives* 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
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/WithUPDATE	
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 *receives* 200 OK final response containing the SDP with the attribute "a=" inactive.	
Precondition:	• A session was established between user A and user B according to the "basic Call" procedures. • The media stream was previously set to "sendonly". • 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
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 *receives* 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 between user A and user B according to the "basic Call" procedures. • The media stream was previously set to "sendonly". • 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.	

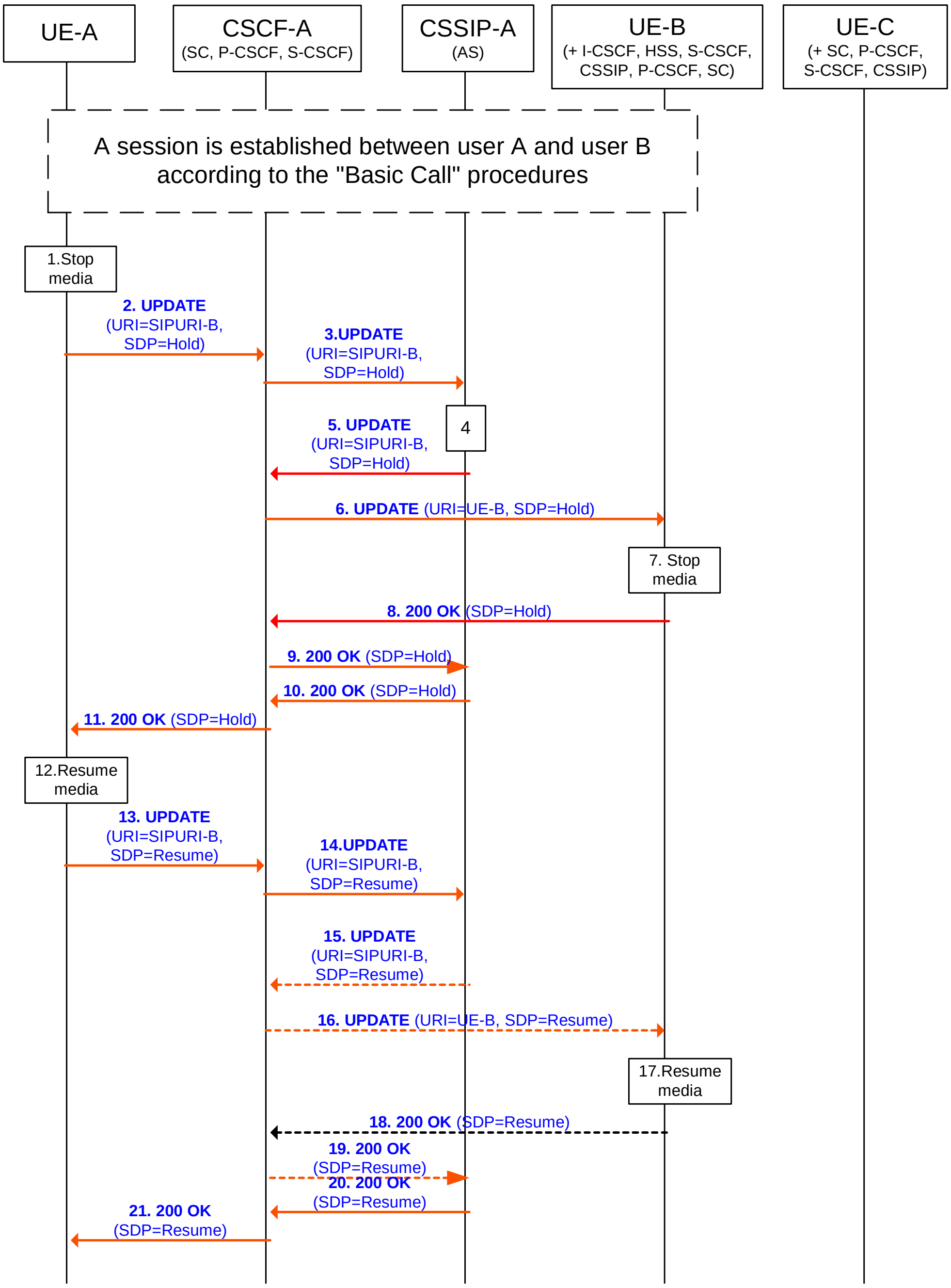
Comments:

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

SS__XXSSCH08	**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:	*All the 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 INVITE to resume the session with the attribute "a=" sendrecv in the SDP. The IUT after requesting the hold session *receives* 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 between user A and user B according to the "basic Call" procedures. • The media stream was previously set to "inactive". • 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
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: 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)	→		→	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/WithoutUPDATE
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 *receives* 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=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**)	→		→	INVITE(sendonly)
200 OK INVITE (**recvonly**)	←		←	200 OK INVITE(recvonly)
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 11	**NGN reference to: [21]**	
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE	
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 *receives* 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=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**)	→		→	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/WithoutUPDATE	
Selection criteria:	*Session hold. INVITE method is used. Individual media streams are affected*	
Test purpose:	*The media stream was previously set to recvonly.* 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 *receives* 200 OK final response containing the SDP with the attribute "a=" inactive.	
Precondition:	• A session was established between user A and user B according to the "basic Call" procedures. • The media stream was previously set to "recvonly". • 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**)	→		→	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

SS__XXSSCH 13	**NGN reference to: [21]**
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE
Selection criteria:	*Session hold. INVITE method is used. Individual media streams are affected.*
Test purpose:	*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 *receives* 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 between user A and user B according to the "basic Call" procedures. • The media stream was previously set to "sendonly". • 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**
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 14	**NGN reference to: [21]**
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE
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 *receives* 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=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)	←		←	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/WithoutUPDATE
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 *receives* 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**)	→		→	INVITE(sendonly)
200 OK INVITE (**recvonly**)	←		←	200 OK INVITE(recvonly)
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 16	**NGN reference to: [21]**	
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE	
Selection criteria:	*Session hold. INVITE method is used. All the media streams.*	
Test purpose:	*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 INVITE to resume the session with the attribute "a=" sendrecv in the SDP. The IUT after requesting the hold session *receives* 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 between user A and user B according to the "basic Call" procedures. • The media stream was previously set to "recvonly". • 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:

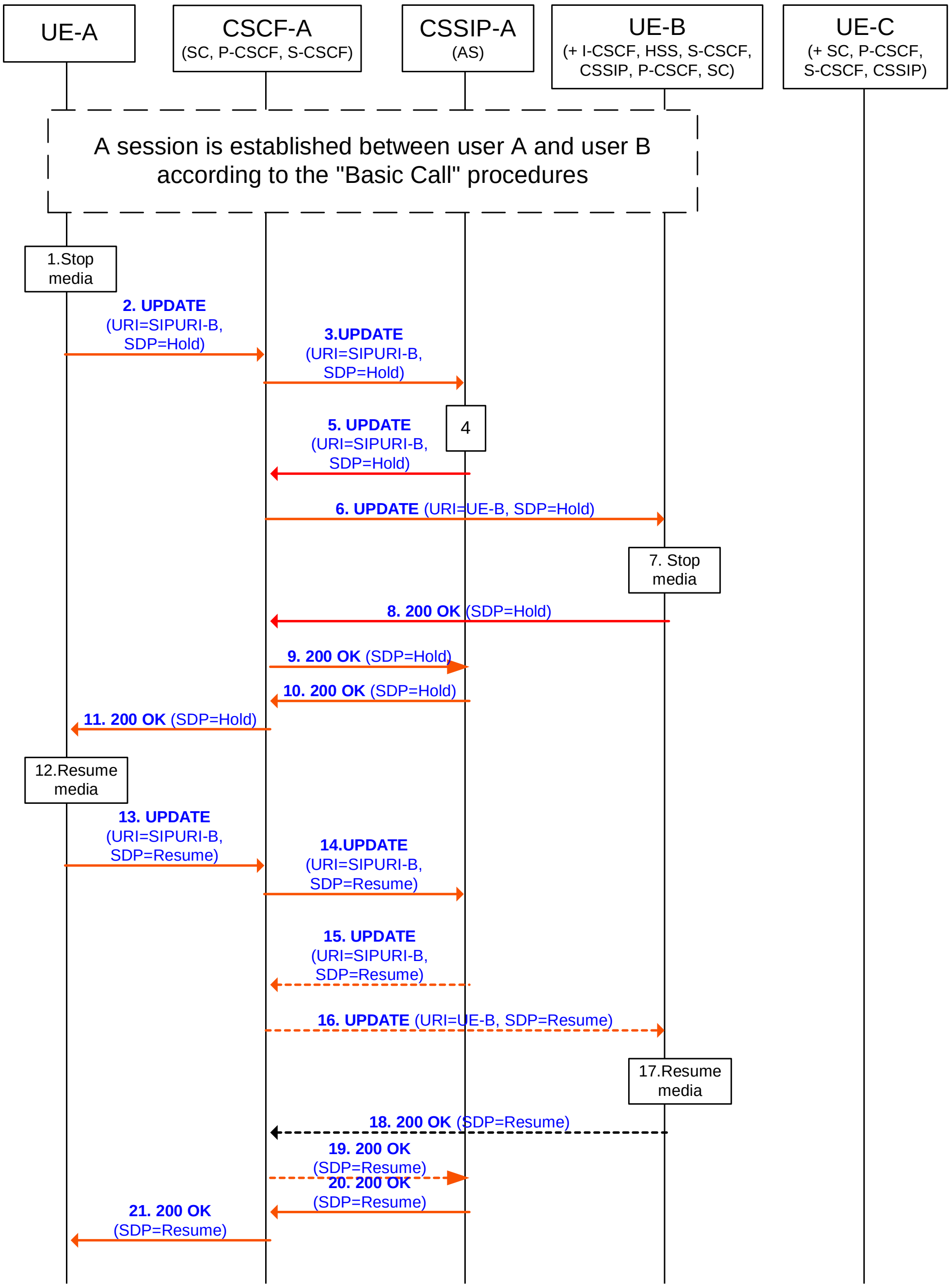
SIP UA A		**SUT**		**SIP UA B**
INVITE (**sendrecv**)	→		→	INVITE
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
INVITE (sendonly)	←		←	INVITE(sendonly)
200 OK INVITE (**recvonly**)	→		→	200 OK INVITE(recvonly)
INVITE(**inactive**)	→		→	INVITE(inactive)
200 OK INVITE (**inactive**)	←	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

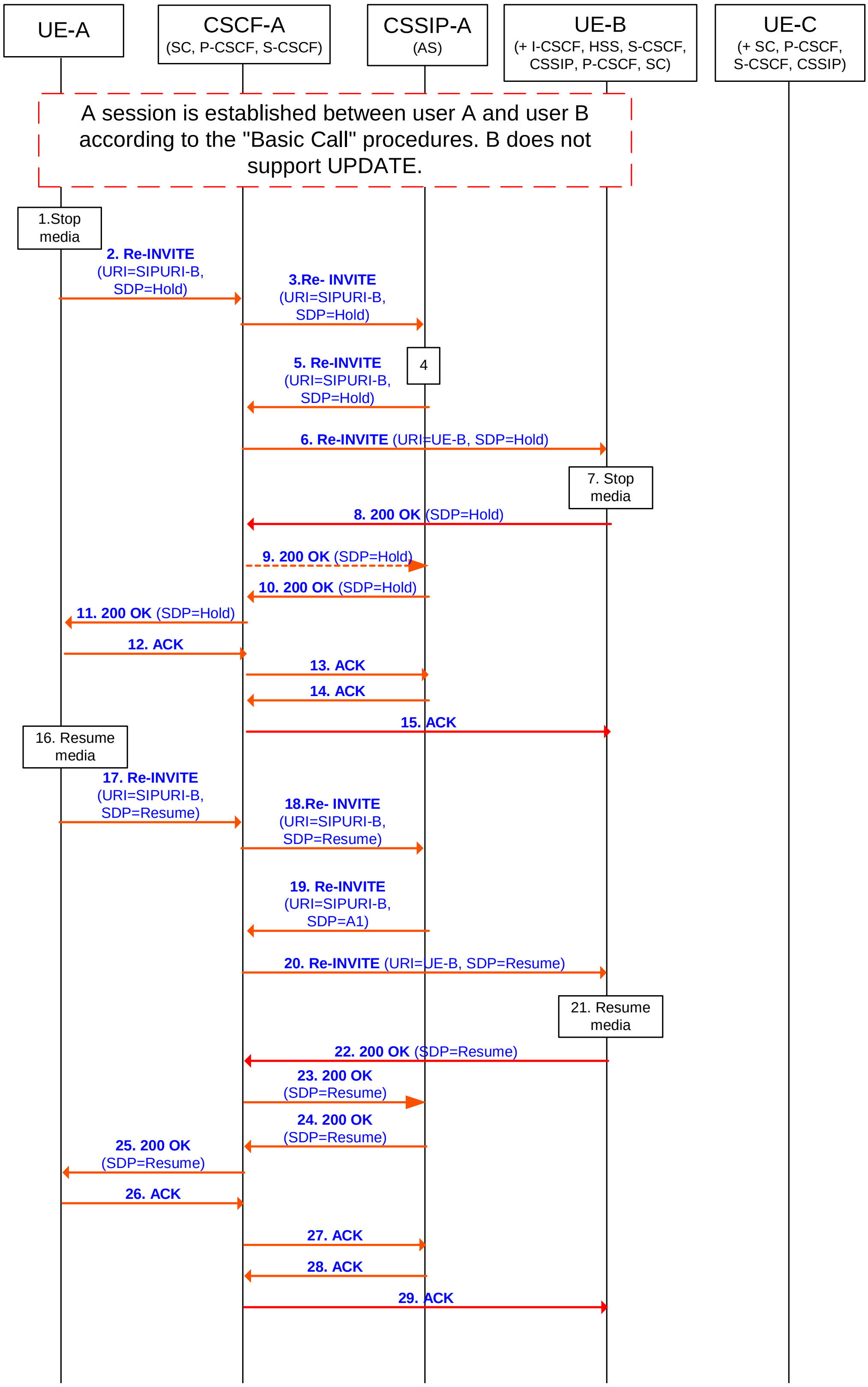
SS__XXSSCH 17	**NGN reference to: [21]**	
TSS reference:	ServedUser/WithoutAnnounc/WithoutUPDATE	
Selection criteria:	*Session hold. INVITE method is used. All the media streams are affected.*	
Test purpose:	*The media stream was previously set to sendonly.* Ensure that the IUT is requesting to **resume** 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 *receives* 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 between user A and user B according to the "basic Call" procedures. • The media stream was previously set to "sendonly". • 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**)	→		→	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/WithoutUPDATE		
Selection criteria:	*Session hold. INVITE method is used. 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 *receives* 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". • 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**
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


UE-A
CSCF-A
(SC, P-CSCF, S-CSCF)
CSSIP-A
(AS)
UE-B
(+ I-CSCF, HSS, S-CSCF, CSSIP, P-CSCF, SC)
UE-C
(+ SC, P-CSCF, S-CSCF, CSSIP)
A session is established between user A and user B according to the "Basic Call" procedures
1.Stop media
2. UPDATE (URI=SIPURI-B, SDP=Hold)
3.UPDATE (URI=SIPURI-B, SDP=Hold)
4
5. UPDATE (URI=SIPURI-B, SDP=Hold)
6. UPDATE (URI=UE-B, SDP=Hold)
7. Stop media
8. 200 OK (SDP=Hold)
9. 200 OK (SDP=Hold)
10. 200 OK (SDP=Hold)
11. 200 OK (SDP=Hold)
12.Resume media
13. UPDATE (URI=SIPURI-B, SDP=Resume)
14.UPDATE (URI=SIPURI-B, SDP=Resume)
15. UPDATE (URI=SIPURI-B, SDP=Resume)
16. UPDATE (URI=UE-B, SDP=Resume)
17.Resume media
18. 200 OK (SDP=Resume)
19. 200 OK (SDP=Resume)
20. 200 OK (SDP=Resume)
21. 200 OK (SDP=Resume)


UE-A
CSCF-A
(SC, P-CSCF, S-CSCF)
CSSIP-A
(AS)
UE-B
(+ I-CSCF, HSS, S-CSCF, CSSIP, P-CSCF, SC)
UE-C
(+ SC, P-CSCF, S-CSCF, CSSIP)
A session is established between user A and user B according to the "Basic Call" procedures. B does not support UPDATE.
1.Stop media
2. Re-INVITE (URI=SIPURI-B, SDP=Hold)
3.Re- INVITE (URI=SIPURI-B, SDP=Hold)
4
5. Re-INVITE (URI=SIPURI-B, SDP=Hold)
6. Re-INVITE (URI=UE-B, SDP=Hold)
7. Stop media
8. 200 OK (SDP=Hold)
9. 200 OK (SDP=Hold)
10. 200 OK (SDP=Hold)
11. 200 OK (SDP=Hold)
12. ACK
13. ACK
14. ACK
15. ACK
16. Resume media
17. Re-INVITE (URI=SIPURI-B, SDP=Resume)
18.Re- INVITE (URI=SIPURI-B, SDP=Resume)
19. Re-INVITE (URI=SIPURI-B, SDP=A1)
20. Re-INVITE (URI=UE-B, SDP=Resume)
21. Resume media
22. 200 OK (SDP=Resume)
23. 200 OK (SDP=Resume)
24. 200 OK (SDP=Resume)
25. 200 OK (SDP=Resume)
26. ACK
27. ACK
28. ACK
29. ACK

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/WithUPDATE
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 set on HOLD, an announcement is started to the remote UE. An UPDATE is sent to the remote 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:

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		
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. 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 UPDATE 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 precondition.

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/WithUPDATE
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:	• 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**)	→		→	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)	←		←	200 OK UPDATE (**sendrecv**)
		Conversation		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 22	NGN reference to: [21]
TSS reference:	ServedUser/WithAnnounc/WithUPDATE
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**)	→		→	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				
UPDATE(recvonly)	←		←	UPDATE(**recvonly**)
200 OK UPDATE (sendonly)	→		→	200 OK UPDATE (**sendonly**)
Announcement to UE B				
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 23	NGN reference to: [21]
TSS reference:	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 streams are affected.*
Test purpose:	Ensure that when the remote user is set on HOLD, an announcement is started to the remote UE. An UPDATE 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=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
Conversation				
UPDATE(sendonly)	→		→	UPDATE(**sendonly**)
200 OK UPDATE (recvonly)	←		←	200 OK UPDATE (**recvonly**)
Announcement to UE B				
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 24	**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. All the media streams are affected.*
Test purpose:	Ensure that no announcement is started 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:	• 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	←		←	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 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	←		←	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/WithAnnounc/WithUPDATE
Selection criteria:	*Announcement is started to user B when user B retrieves the connection. All the 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. • 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
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**)
		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=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)	→		→	INVITE (**sendonly**)
200 OK INVITE (recvonly)	←		←	200 OK INVITE (**recvonly**)
		Announcement to UE B		
BYE	→		→	BYE
200 OK 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 precondition.

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)
		Announcement to UE B		
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**)	→		→	INVITE
180 Ringing	←		←	180 Ringing
200 OK INVITE	←		←	200 OK INVITE
INVITE(sendonly)	→		→	INVITE(sendonly)
200 OK INVITE (recvonly)	←		←	200 OK INVITE (**recvonly**)
		Announcement to UE B		
INVITE (sendrecv)	→		→	INVITE (**sendrecv**)
200 OK INVITE (sendrecv)	←		←	200 OK INVITE (**sendrecv**)
		Conversation		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 30	NGN reference to: [21]	
TSS reference:	ServedUser/WithAnnounc/WithoutUPDATE	
Selection criteria:	*Announcement is started to user B when user B retrieves the connection. The INVITE method is used. 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 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. • 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**)	→		→	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**)
		Announcement to UE B		
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

SS__XXSSCH 31	**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. 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=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)	→		→	INVITE (**sendonly**)
200 OK INVITE (recvonly)	←		←	200 OK INVITE (**recvonly**)
		Announcement to UE B		
BYE	→		→	BYE
200 OK 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 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)	→		→	INVITE (**sendonly**)
200 OK INVITE (recvonly)	←		←	200 OK INVITE(recvonly)
		Announcement to UE B		
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 33	**NGN reference to: [21]**	
TSS reference:	ServedUser/WithAnnounc/WithoutUPDATE	
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:	• 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	←	←	180 Ringing
200 OK INVITE	←	←	200 OK INVITE
INVITE(sendonly)	→	→	INVITE(sendonly)
200 OK INVITE (recvonly)	←	←	200 OK INVITE (**recvonly**)
	Announcement to UE B		
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]
TSS reference:	ServedUser/WithAnnounc/WithoutUPDATE
Selection criteria:	*Announcement is started to user B when user B retrieves the connection. The INVIT method id used. All the 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 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.
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)	→		→	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**)
		Announcement to UE B		
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:	*The caller puts the media stream on hold; the early dialogue is established.* Ensure that the IUT requesting to **hold the session** 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 *receives a 200 OK (UPDATE) message* containing the SDP with the attribute "a=" **reconly**.
Precondition:	Early dialogue was established between user A and user B according to the "basic Call" procedures.
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
UPDATE(**sendonly**)	→		→	UPDATE(sendonly)
200 OK UPDATE (**recvonly**)	←		←	200 OK UPDATE (recvonly)
BYE	→		→	BYE
200 OK BYE	←		←	200 OK BYE

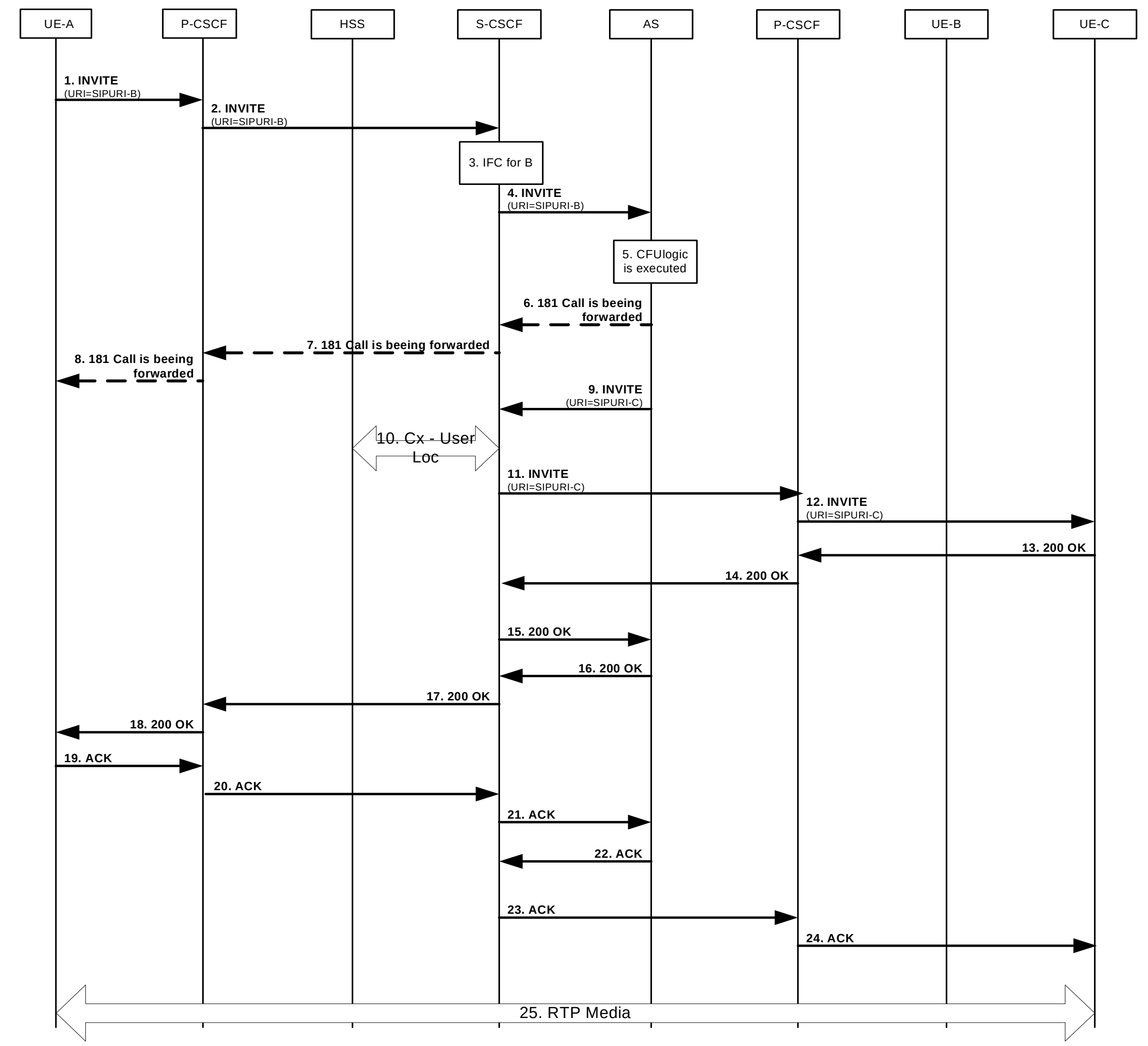
SS__XXSSCH 34	**NGN reference to: [21]**	
TSS reference:	ServedUser/RingingState	
Selection criteria:		
Test purpose:	*The caller retrieves the media stream; the early dialogue is established.* Ensure that the IUT is requesting to **resume the session** with user B the UE-A starts sending media and sends 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 user A and user B according to the "basic Call" procedures	
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:

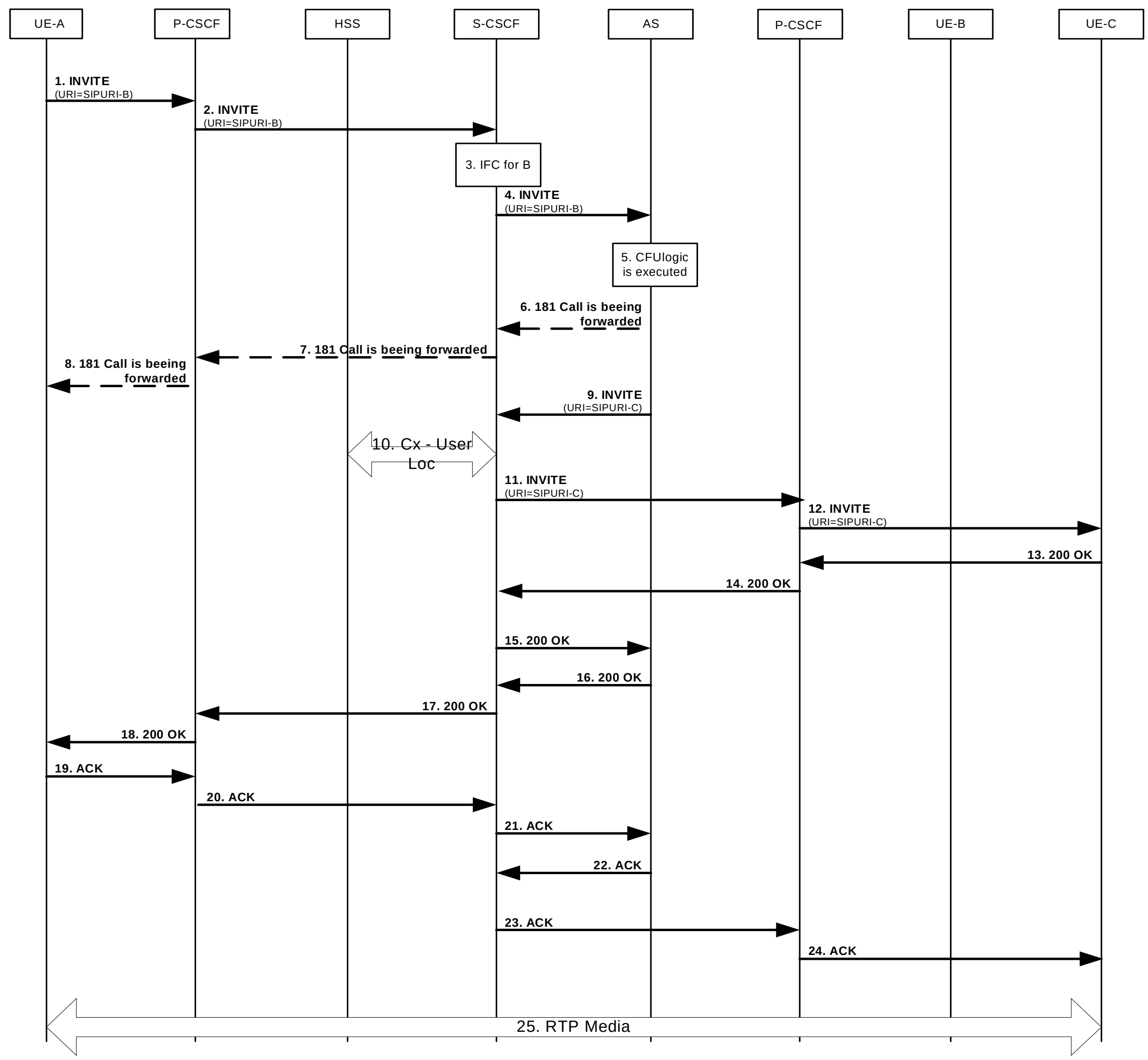
SIP UA A		**SUT**		**SIP UA B**
INVITE (**sendrecv**)	→		→	INVITE
180 Ringing	←		←	180 Ringing
UPDATE(**sendonly**)	→		→	UPDATE(sendonly)
200 OK UPDATE (**recvonly**)	←		←	200 OK UPDATE (recvonly)
UPDATE(**sendrecv**)	→		→	UPDATE(sendrecv)
200 OK UPDATE (**sendrecv**)	←		←	200 OK UPDATE (sendrecv)
BYE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

6.5.2.6 CFU

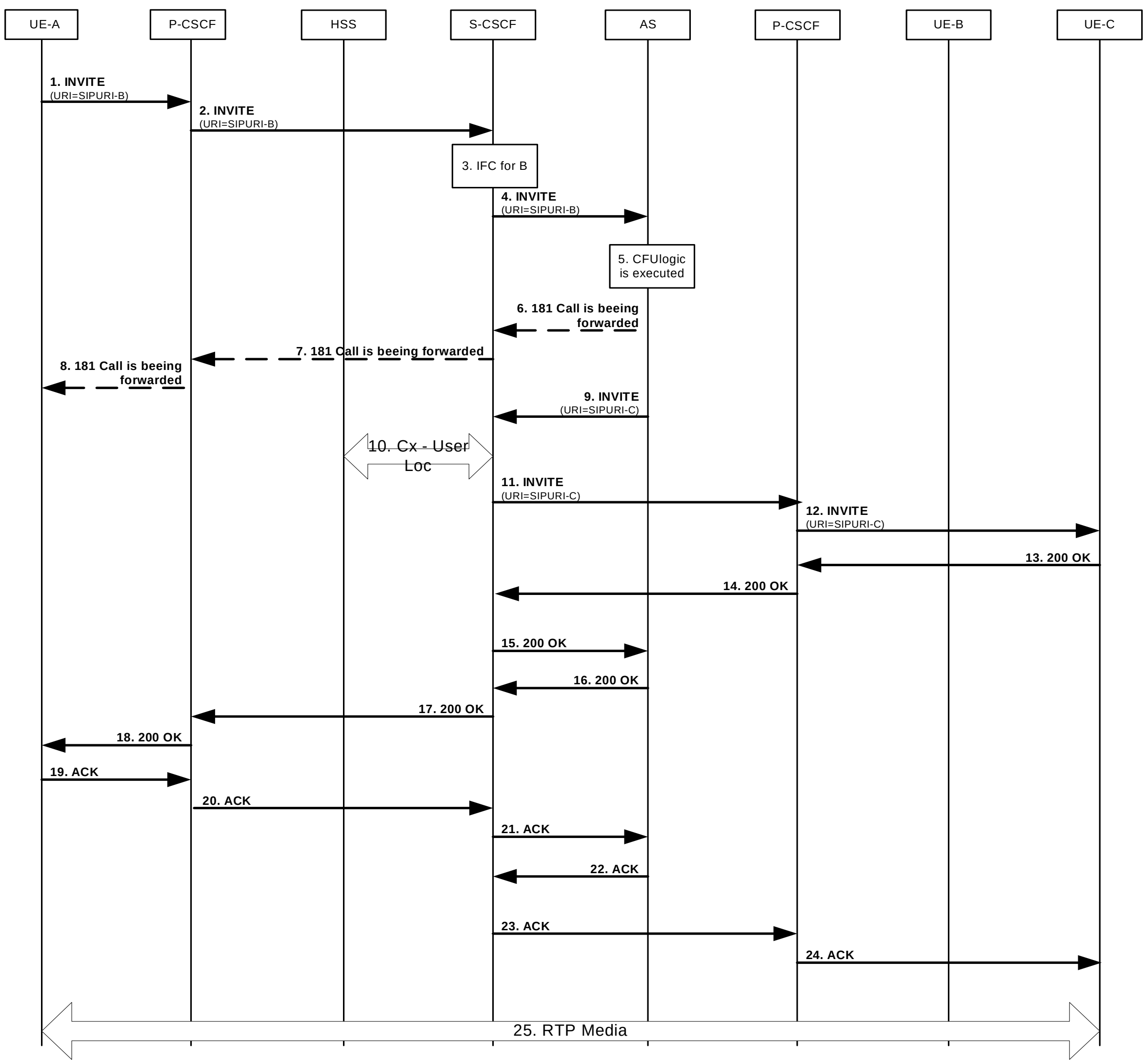
SSS__XXSSCFU 01	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFU
Configuration	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU.
Selection criteria:	• Call forwarding by the network. • Call forwarding unconditional supported.
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters).
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:	



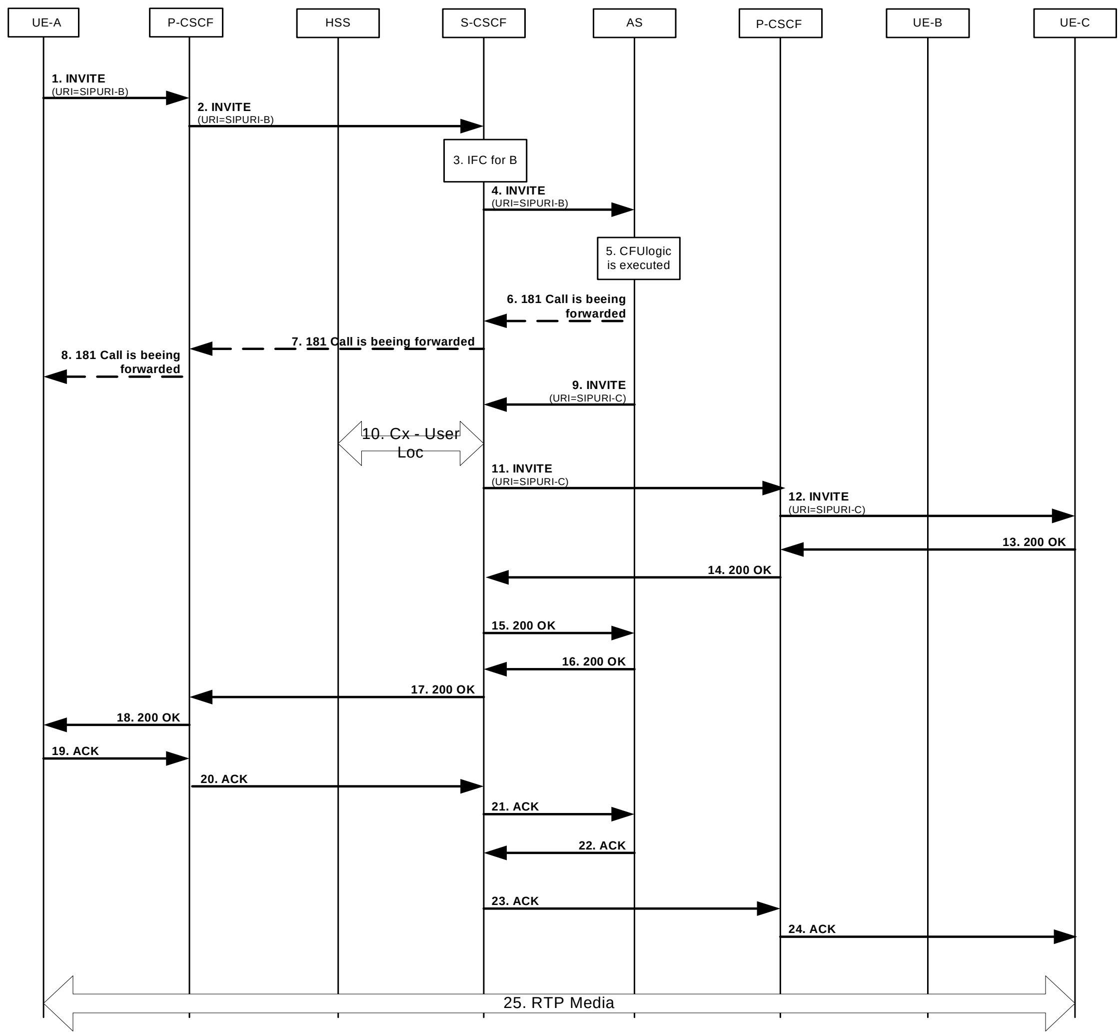
SSS__XXSSCFU 02	**NGN reference to: [17]**
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFU
Configuration	The user A and the user C are in network N1. The user B is provided with CFU. ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting number is released to the diverted-to user"=Yes, "served user receives notification that the call has been forwarded" =Yes).
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 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:	



SSS__XXSSCFU 03	**NGN reference to: [17]**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFU	
Configuration	The user A and the user C are in network N1. The user B is provided with CFU. ("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).	
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 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:		

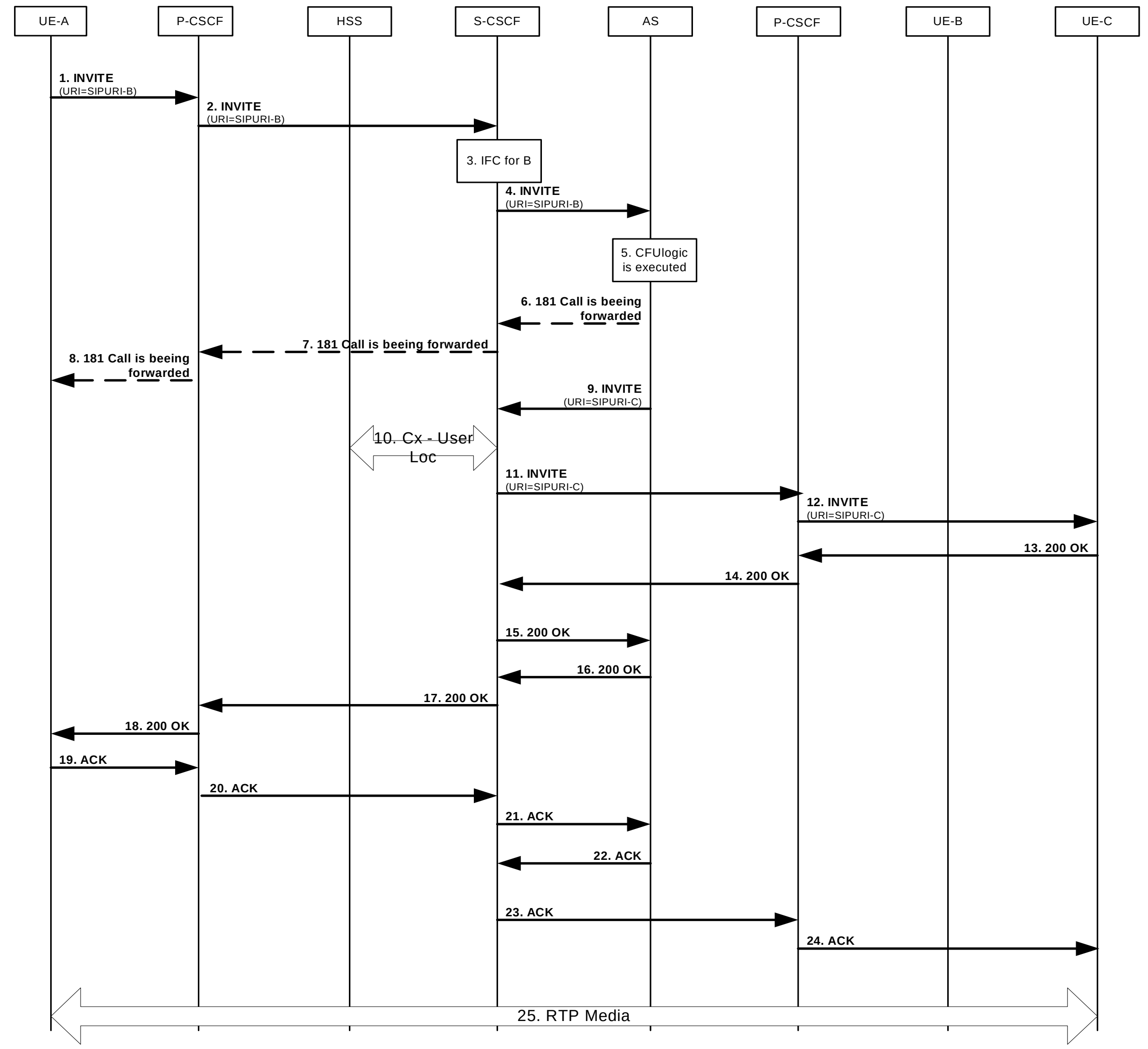


SSS__XXSSCFU 04	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFU
Configuration	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFU ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "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 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:	

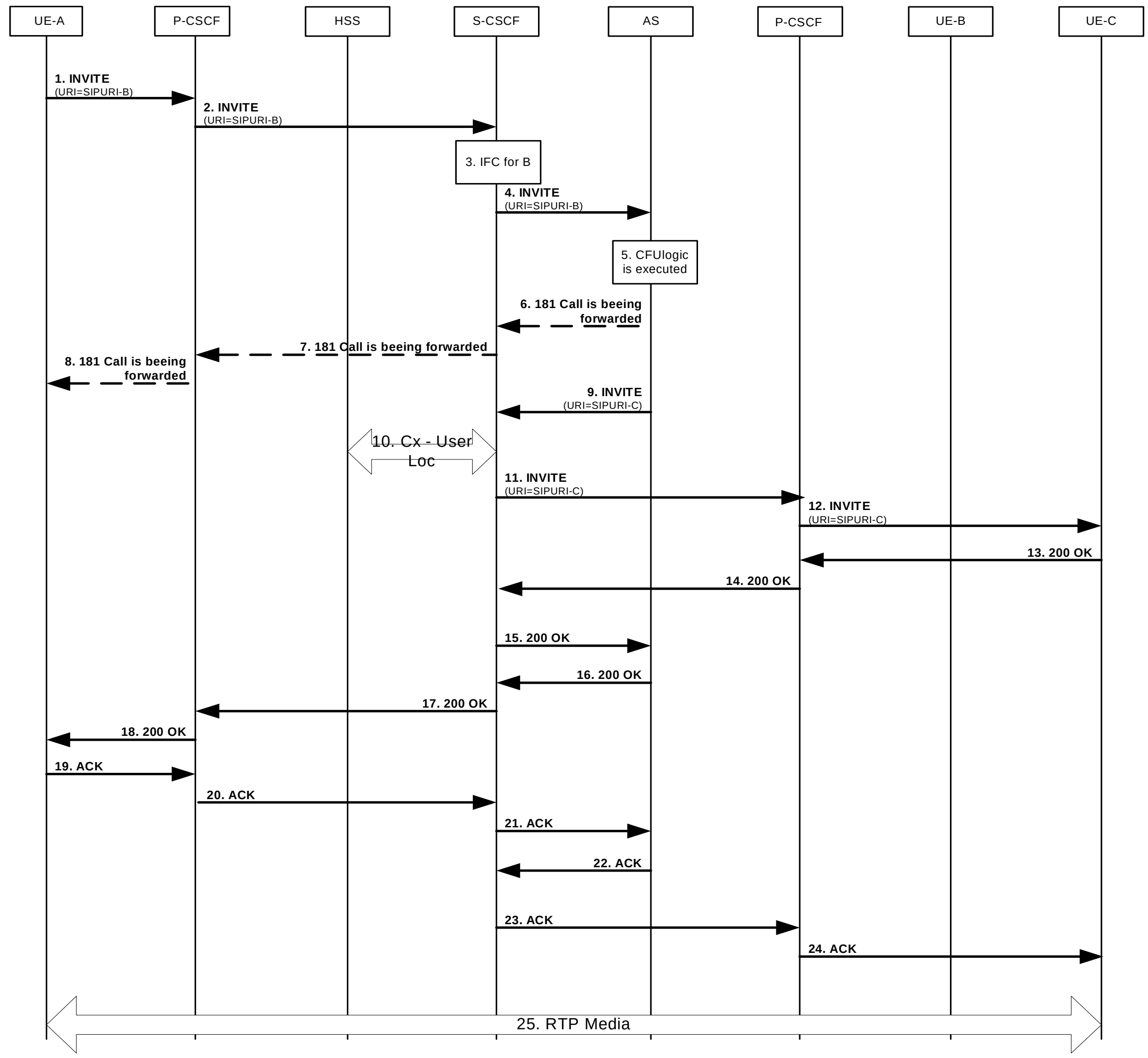


SSS__XXSSCFU 05	NGN reference to: [17]	
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFU	
Configuration	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFU. The user A and the user C are in network N1 and user C is provided with TIP. The user B 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.	

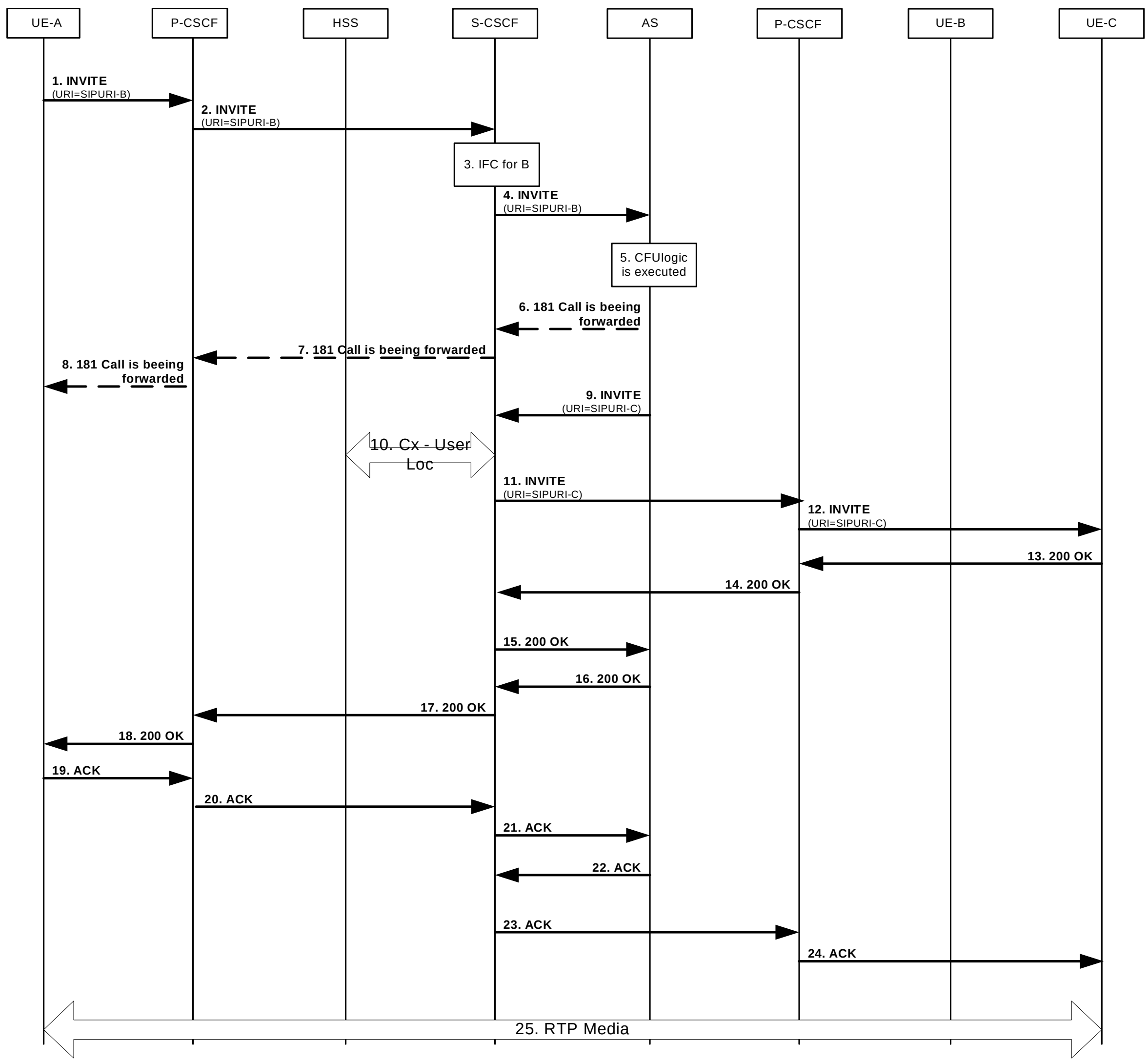
SSS__XXSSCFU 05	NGN reference to: [17]	
Comments:		



SSS__XXSSCFU 06	**NGN reference to: [17]**
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFU
Configuration	The user B is in network N2 and is provided with CFU
Selection criteria:	• Call forwarding by the network. • Call forwarding uncontitional supported. • user C is user determined user busy.
Test purpose:	To verify that a call is released correctly if **CFU 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__XXSSCFU 07	**NGN reference to: [17]**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFU	
Configuration	The user B is in network N2 and is provided with CFU	
Selection criteria:	• Call forwarding by the network • Call forwarding uncontitional supported • user C is network determined user busy	
Test purpose:	To verify that a call is released correctly if CFU 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__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 provided with CFU.	
Selection criteria:	• Call forwarding by the network. • Call forwarding uncontitional supported. • user C is network determined user busy.	
Test purpose:	*The served user subscribes to the CFU simulation service; originating user is notified. Additionaly an announcement is played.* When Communication Diversion occurs and if the notification procedures of the originating user is supported then a 181 (Call Is Being Forwarded) response shall be sent towards the originating user. Additional the AS initiates an announcement to be included towards the calling user in order to inform about the diversion. Announcements shall be played according to procedures as are described in TS 183 028 [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-Asserted-Identity SIP#2.	

Comments:

SIP#1		**SUT**	**SIP#2 (served user)**	**SIP#3**
INVITE 1	→			
		Communication diversion is performed		
181 Call is Being Forwarded	←	181 Call is Being Forwarded		
		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
		Communication		
BYE	→			→ BYE
200 OK (BYE)	←			← 200 OK (BYE)

The test case can not 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/CFU	
Configuration	The user B is in network N2 and is provided with CFU	
Selection criteria:	• Call forwarding by the network • Call forwarding uncontitional supported • user C is network determined user busy	
Test purpose:	*The served user subscribes to the CFU simulation service; Indication of communication diversion to the diverting user using the MESSAGE request triggered by a timer.* Ensure that when the diverting user is registering the AS sends a MESSAGE request including the information where the call is diverted to. The MESSAGE request that is be sent due to an timer value that can be provided by the 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. MESSAGE triggerd by a timer value provided the served user.	

Comments:

SIP#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→					
		Communication diversion is performed				
		INVITE 2	→		→	INVITE
		MESSAGE	→	MESSAGE		
		200 OK MESSAGE		200 OK MESSAGE		
		180 Ringing	←		←	180 Ringing
180 Ringing	←	180 Ringing				
		200 OK (INVITE)	←		←	200 OK (INVITE)
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)

The test case can not be tested with an ATS

SSS__XXSSCFU 10	**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 provided with CFU	
Selection criteria:	• Call forwarding by the network. • Call forwarding uncontitional supported. • user C is network determined user busy.	
Test purpose:	*The served user subscribes to the CFU simulation service; Indication of communication diversion to the diverting user using the MESSAGE request when a new outgoing communication is requested.* Ensure that a diverting user will be informed with a MESSAGE request after the diverting user has initiated a new outgoing communication the information where the 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; Case c) Supported: 100 rel and precondition.	

Comments:

SIP#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→					
Communication diversion is performed						
		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
			→	INVITE 3	→	INVITE
			→	MESSAGE		
				200 OK MESSAGE		
			←	180 Ringing	←	180 Ringing
			←	200 OK (INVITE)	←	200 OK (INVITE)
			→	ACK	→	ACK

The test case can not be tested with an ATS

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 provided with CFU.	
Selection criteria:	• Call forwarding by the network. • Call forwarding uncontitional supported. • user C is network determined user busy.	
Test purpose:	*The served user subscribes to the CFU simulation service; Indication of communication diversion to the diverting user using a voicemail or mail system.* Ensure that when communication diversion is performed, the AS sends a **Voicemail or Message mail system** request including the information where the 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; Case c) Supported: 100 rel and precondition.	

Comments:

SIP#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→					
		Communication 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	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)

The test case can not be tested with an ATS

SSS__XXSSCFU 12	**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 provided with CFU.	
Selection criteria:	• Call forwarding by the network. • Call forwarding uncontitional supported. • user C is network determined user busy.	
Test purpose:	*The served user subscribes to the CFU simulation service; Indication of communication diversion to the diverting user using a voicemail or mail system requested by a timer.* Ensure that when communication diversion is performed, the AS sends a Voicemail or Message mail system request including the information where the call is diverted to. The Voicemail or Message mail system request that is be sent due to a timer value that can be provided by the 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#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→					
		Communication 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	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)
The test case can not be tested with an ATS						

SSS__XXSSCFU 13	**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 provided with CFU.			
Selection criteria:	• Call forwarding by the network. • Call forwarding uncontitional supported. • user C is network determined user busy.			
Test purpose:	*The served user subscribes to the CFU simulation service; periodically indication of communication diversion to the diverting user using a voicemail or mail system.* Ensure that when communication diversion is performed. The AS sends a Voicemail or Message mail system request including the information where the call is diverted to. The diverting user will be informed periodically with a Voicemail or Message mail system request the information where the 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; Case c) Supported: 100 rel and precondition.			
Comments:				
SIP#1		**SUT**	**SIP#2 (served user)**	**SIP#3**
INVITE 1	→			
	Communication 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	→			→ ACK
			Voicemail or Message mail system	
		Communication		
BYE	→			→ BYE
200 OK (BYE)	←			← 200 OK (BYE)
The test case can not be tested with an ATS				

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TS 183 004 [17] clause 4.5.2.6.3</th><th></th></tr>
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<tr><td>Configuration</td><td colspan="2">The user B is in network N2 and is provided with CFU</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">• Call forwarding by the network
• Call forwarding uncontitional supported
• user C is network determined user busy</td></tr>
<tr><td>Test purpose:</td><td colspan="2">The served user subscribes to the CFU simulation service; Indication of a new communication diversion to the diverting user using a voicemail or mail system.

Ensure that a diverting user will be informed with a Voicemail or Message mail system request after the diverting user has initiated a new outgoing communication the information where the call is diverted to.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">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</td></tr>
<tr><td colspan="3">Comments:
<table>
<tr><th>SIP#1</th><th></th><th>SUT</th><th></th><th>SIP#2 (served user)</th><th></th><th>SIP#3</th></tr>
<tr><td>INVITE 1</td><td>→</td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td colspan="7">Communication diversion is performed</td></tr>
<tr><td></td><td></td><td>INVITE 2</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>Voicemail or Message mail system</td><td></td><td></td></tr>
<tr><td></td><td></td><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>180 Ringing</td><td>←</td><td>180 Ringing</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>200 OK (INVITE)</td><td>←</td><td></td><td>←</td><td>200 OK (INVITE)</td></tr>
<tr><td>200 OK (INVITE)</td><td>←</td><td>200 OK (INVITE)</td><td></td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td colspan="7">Communication</td></tr>
<tr><td>BYE</td><td>→</td><td></td><td></td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>200 OK (BYE)</td><td>←</td><td></td><td></td><td></td><td>←</td><td>200 OK (BYE)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>INVITE 3</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>Voicemail or Message mail system</td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>180 Ringing
180 Ringing</td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK (INVITE)</td><td>←</td><td>200 OK (INVITE)</td></tr>
</table>
The test case can not be tested with an ATS</td></tr>
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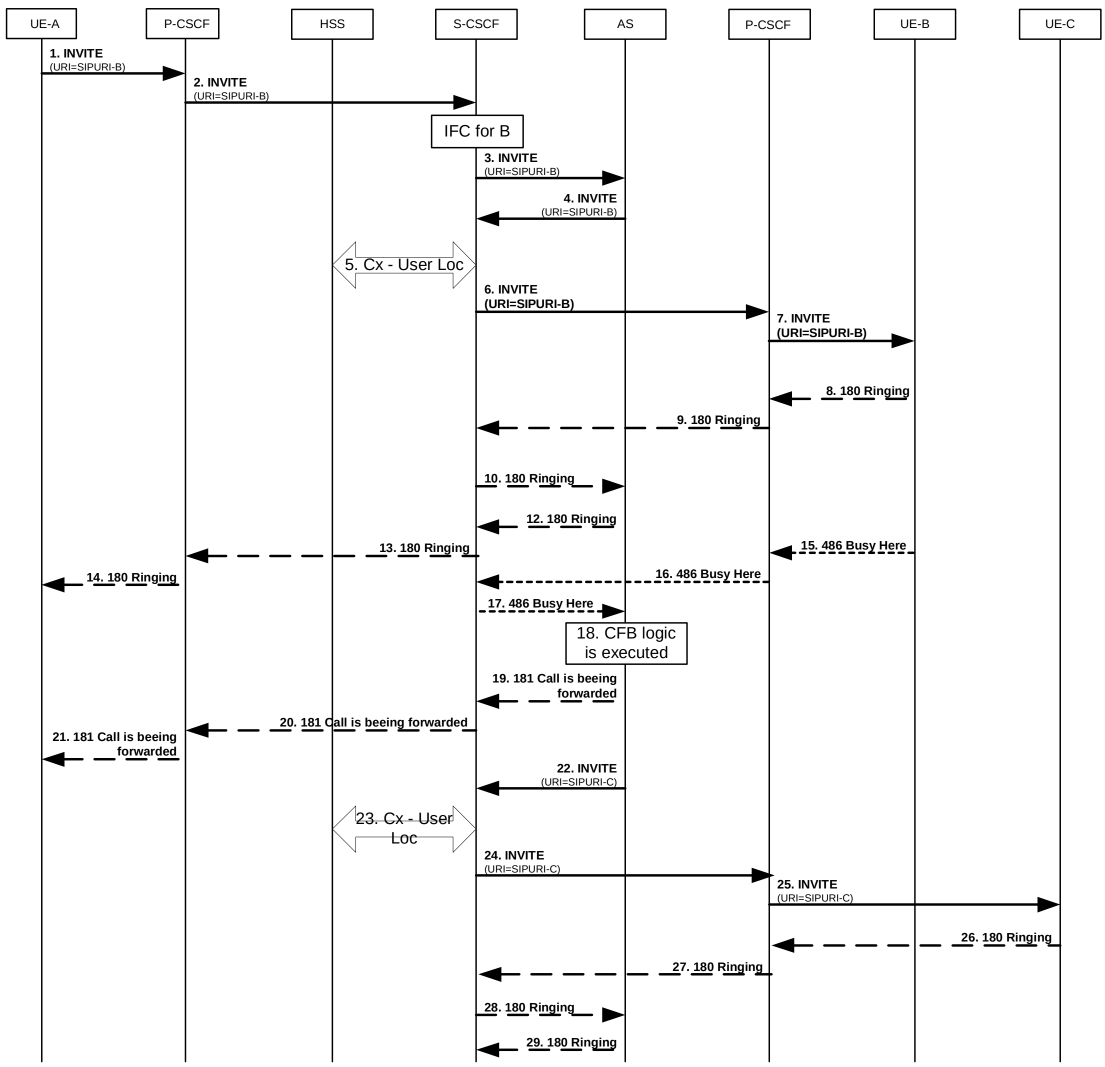
6.5.2.7 CFB

SSS__XXSSCFB 01	**NGN reference to: [17]**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFB	
Configuration	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB (user determined).	
Selection criteria:	• Call forwarding by the network. • Call forwarding busy supported.	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters).	
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:		



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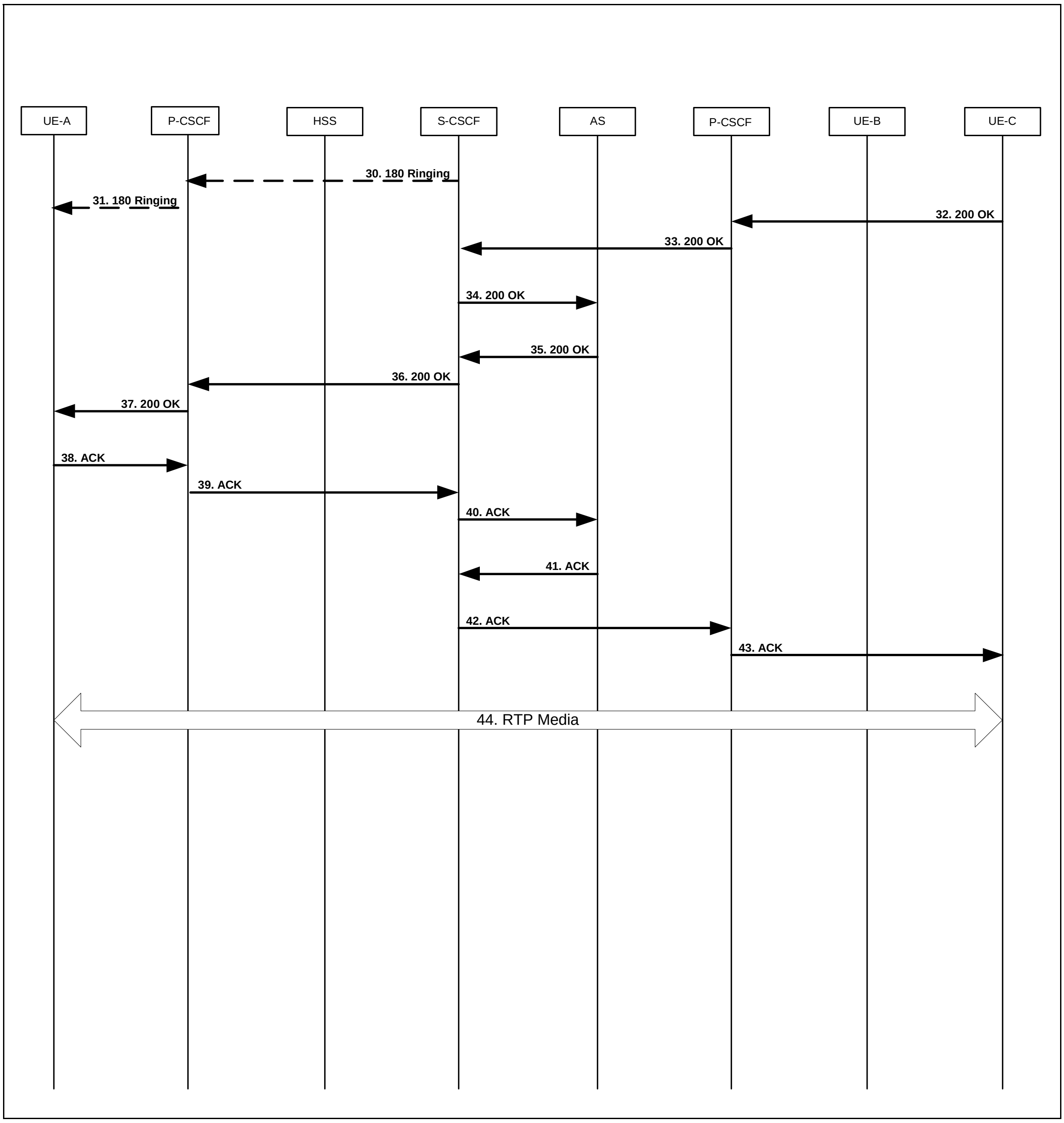
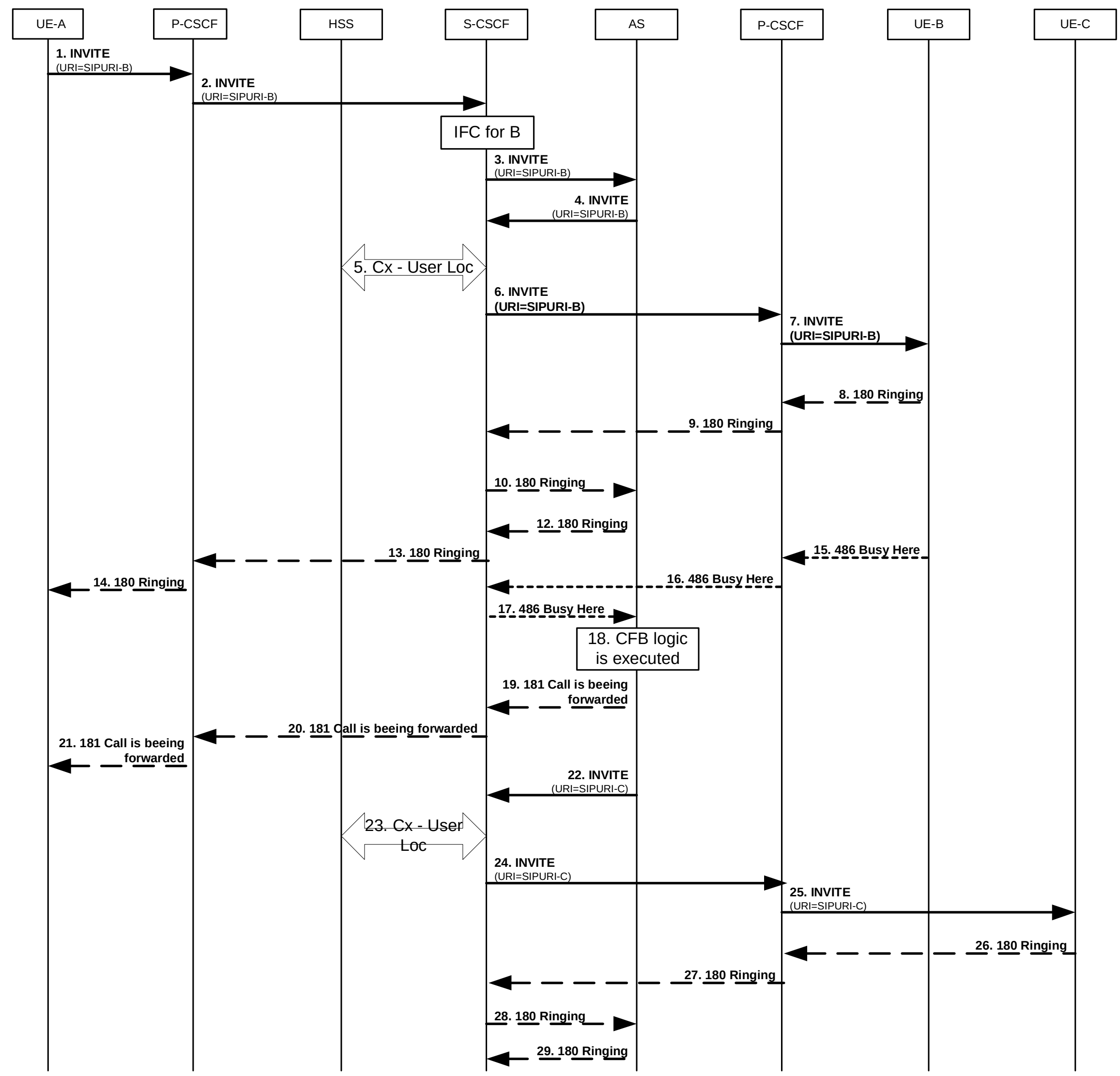

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
30. 180 Ringing
31. 180 Ringing
32. 200 OK
33. 200 OK
34. 200 OK
35. 200 OK
36. 200 OK
37. 200 OK
38. ACK
39. ACK
40. ACK
41. ACK
42. ACK
43. ACK
44. RTP Media

SSS__XXSSCFB 02	**NGN reference to: [17]**
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFB
Configuration	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFB (networtk determined)
Selection criteria:	• Call forwarding by the network • Call forwarding busy supported
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C.
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:



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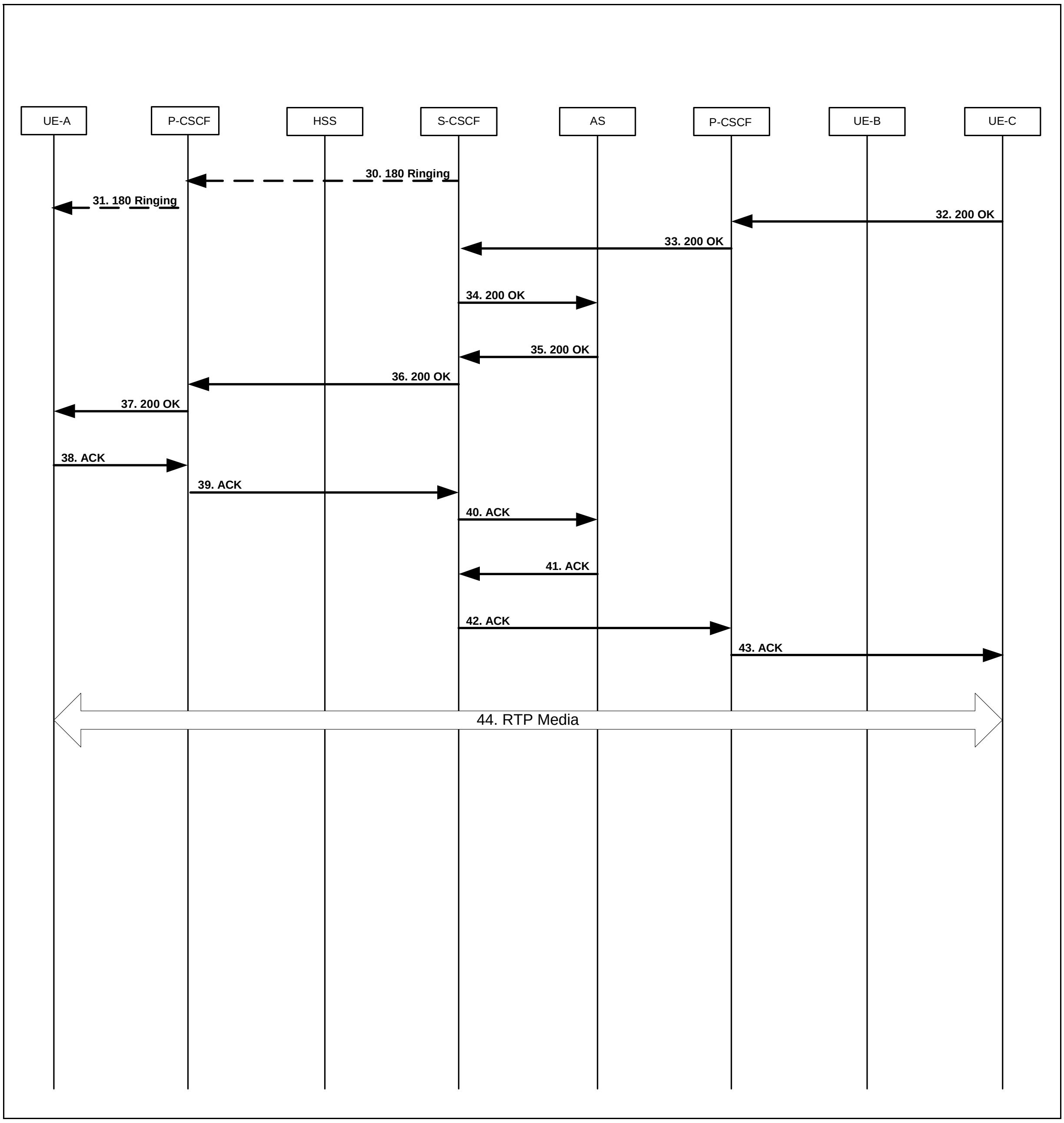
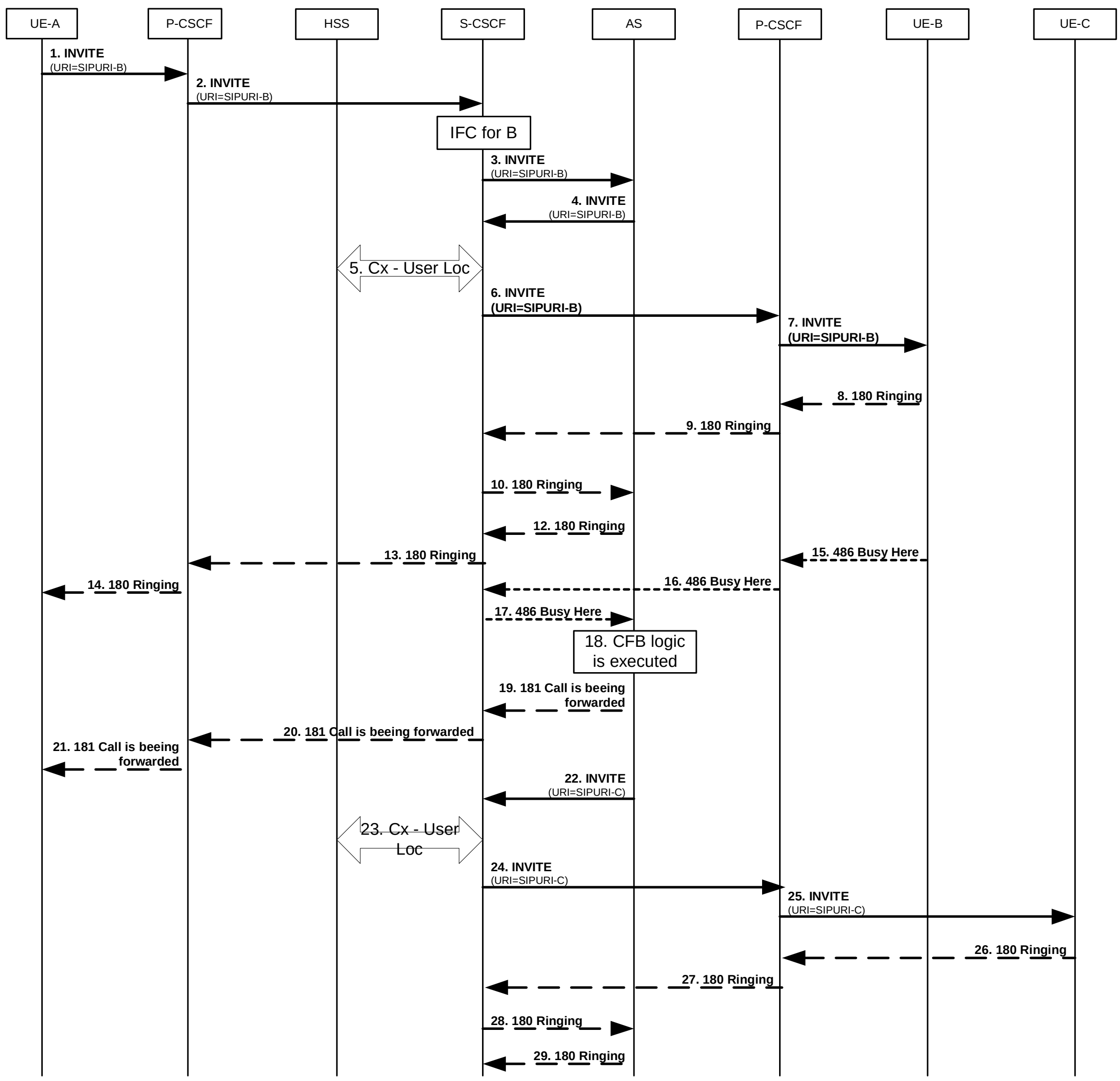

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
30. 180 Ringing
31. 180 Ringing
32. 200 OK
33. 200 OK
34. 200 OK
35. 200 OK
36. 200 OK
37. 200 OK
38. ACK
39. ACK
40. ACK
41. ACK
42. ACK
43. ACK
44. RTP Media

SSS__XXSSCFB 03	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFB
Configuration	The user A and the user C are in network N1. The user B is provided with CFB. ("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).
Selection criteria:	• Call forwarding by the network • Call forwarding busy supported (user determined) • 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 informed of the diverted-to number (user C has TIP) 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:



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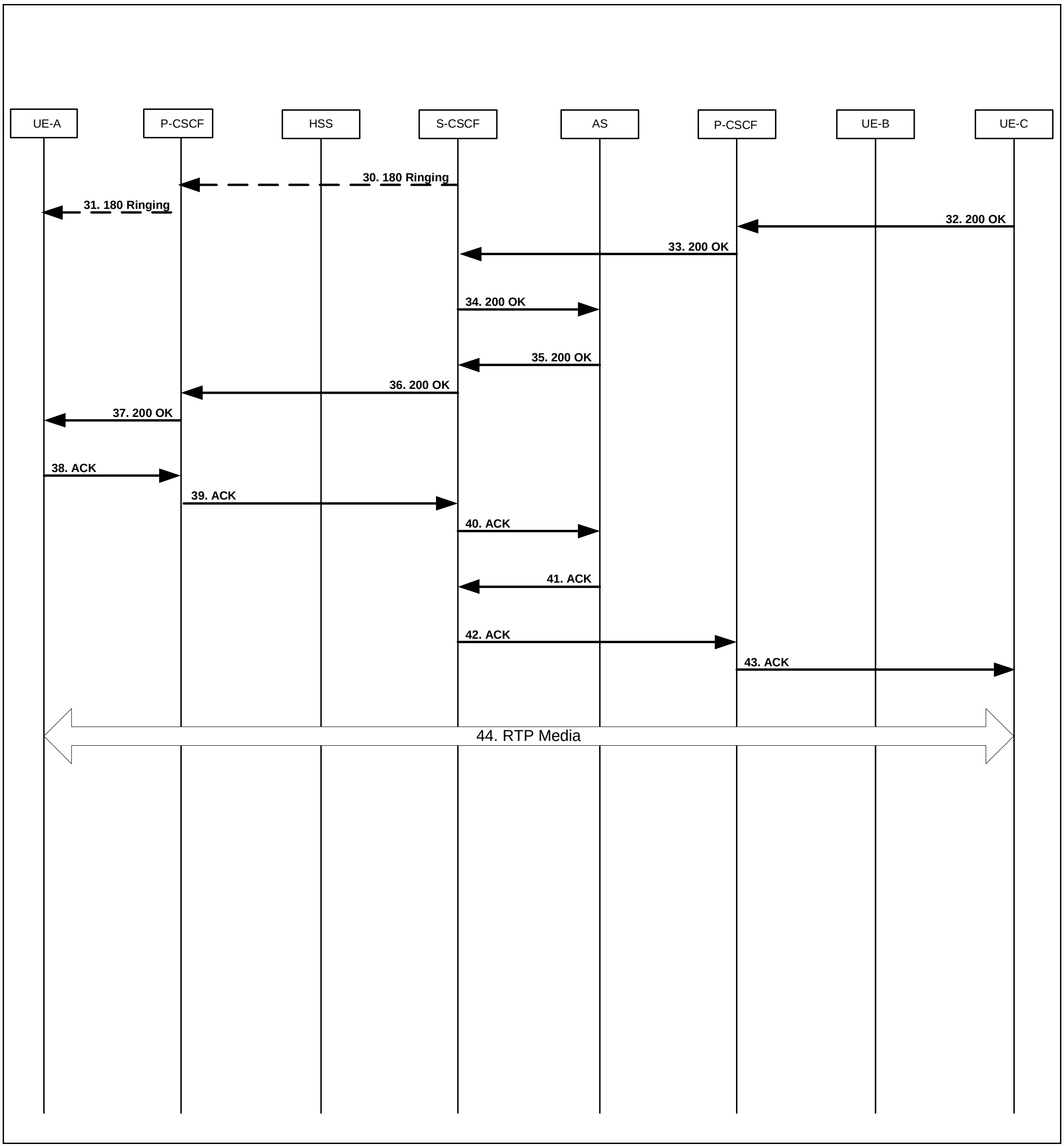
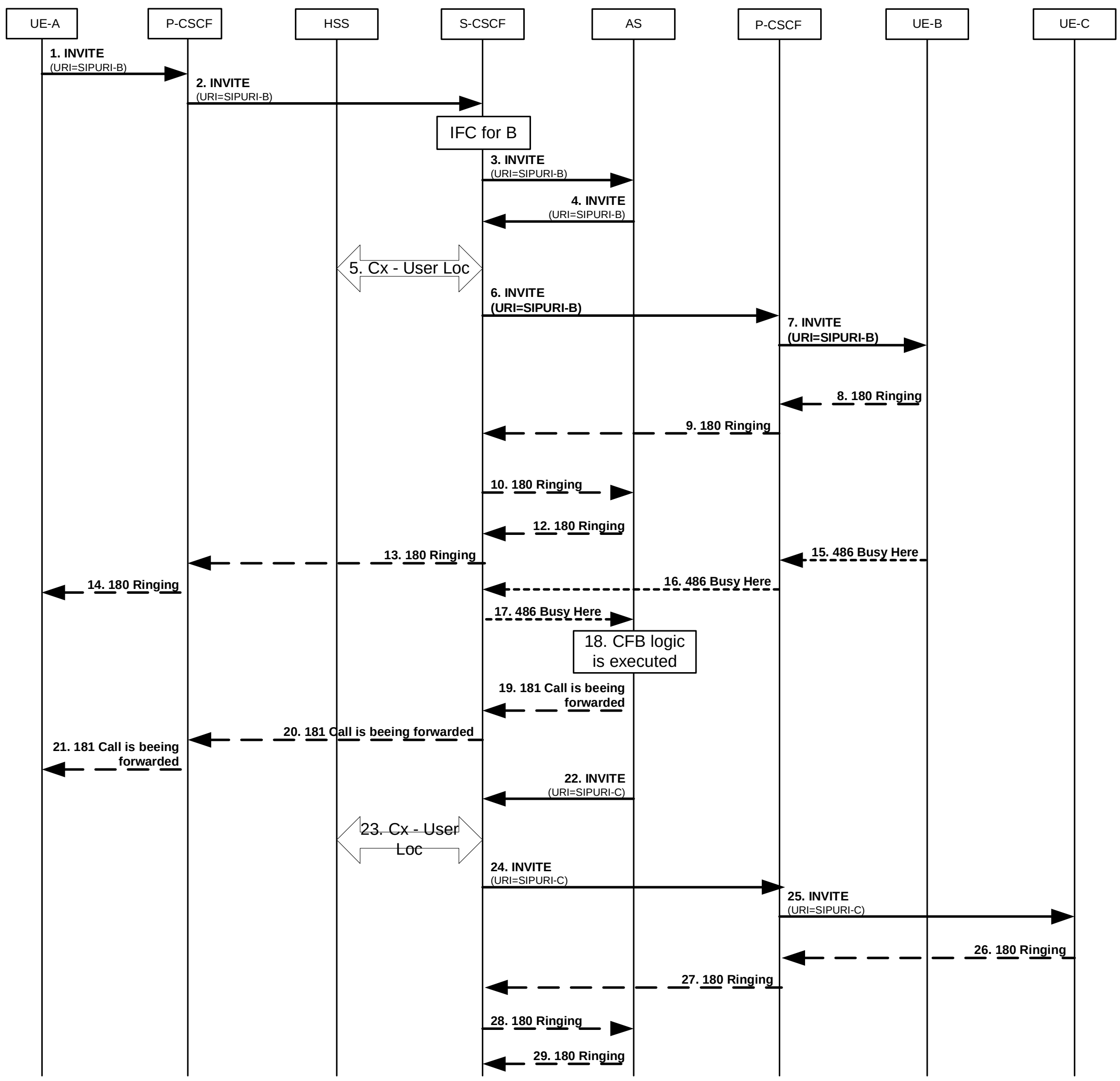


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
30. 180 Ringing
31. 180 Ringing
32. 200 OK
33. 200 OK
34. 200 OK
35. 200 OK
36. 200 OK
37. 200 OK
38. ACK
39. ACK
40. ACK
41. ACK
42. ACK
43. ACK
44. RTP Media

SSS__XXSSCFB 04	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFB
Configuration	The user A and the user C are in network N1. The user B is provided with CFB. ("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).
Selection criteria:	• Call forwarding by the network. • Call forwarding busy supported (user determined). • 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 informed of the diverted-to number (user C has TIP) 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:



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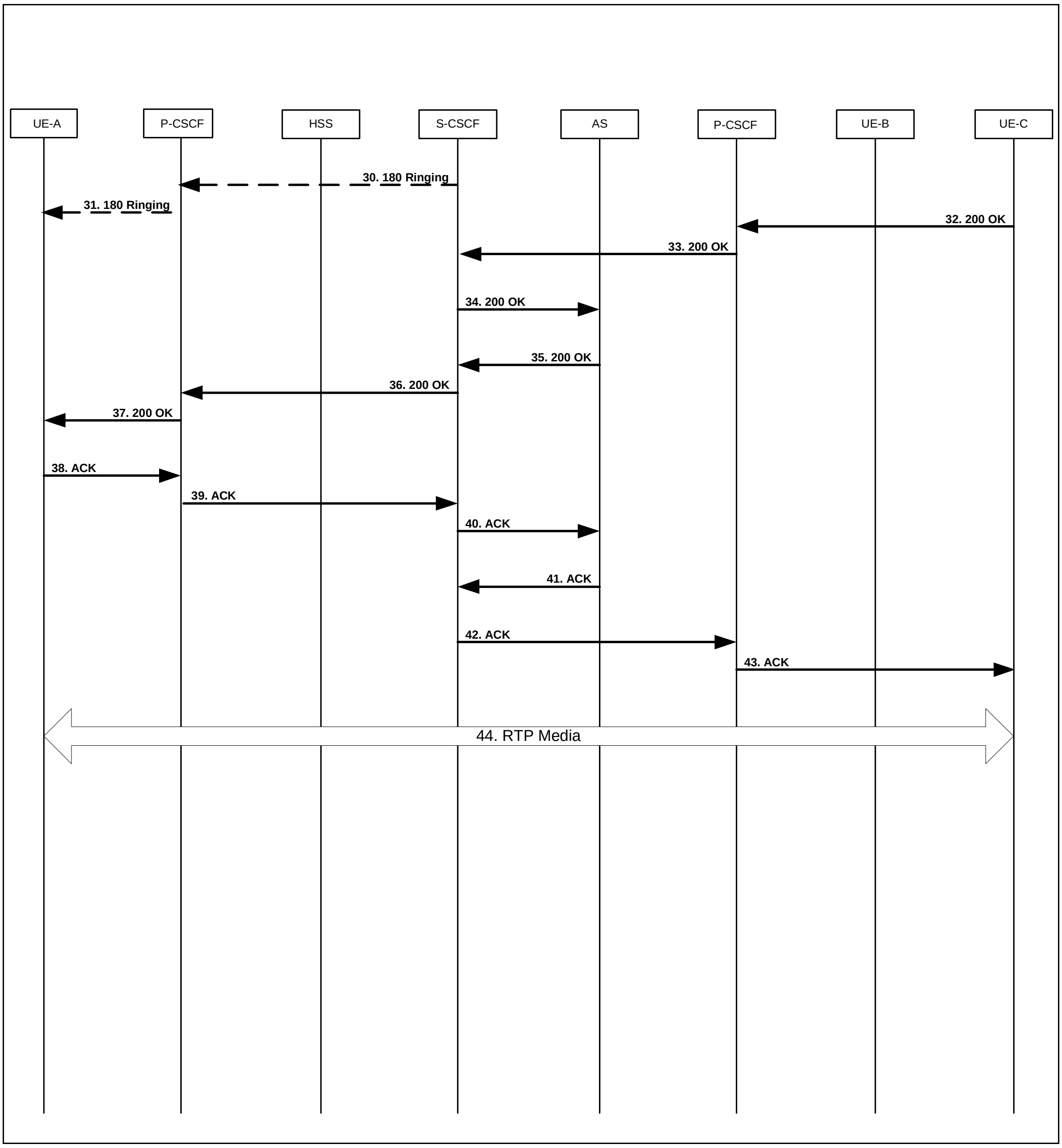
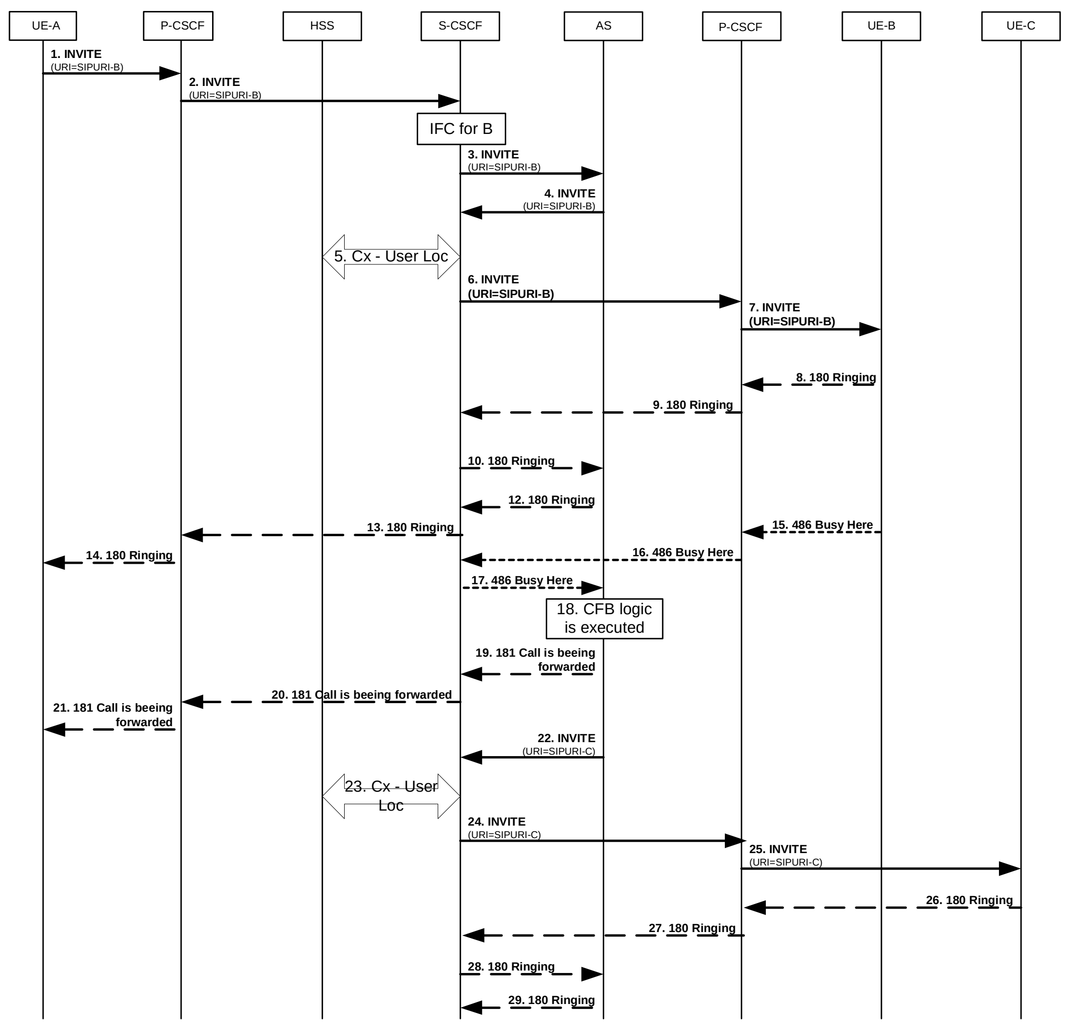


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
30. 180 Ringing
31. 180 Ringing
32. 200 OK
33. 200 OK
34. 200 OK
35. 200 OK
36. 200 OK
37. 200 OK
38. ACK
39. ACK
40. ACK
41. ACK
42. ACK
43. ACK
44. RTP Media

<table>
<tr><th>SSS__XXSSCFB 05</th><th>NGN reference to: [17]</th><th></th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-SIP-SIP/Supplementary_services/CFB.</td></tr>
<tr><td>Configuration</td><td colspan="2">The user A and the user C are in network N1. 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"=Yes, "diverting number is released to the diverted-to user"=No, "served user receives notification that the call has been forwarded" =no).</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">• Call forwarding by the network.
• Call forwarding Busy supported (user determined).
• CF Notifications supported.</td></tr>
<tr><td>Test purpose:</td><td colspan="2">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.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">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.</td></tr>
<tr><td colspan="3">Comments:

</td></tr>
<tr><td colspan="3">Continued</td></tr>
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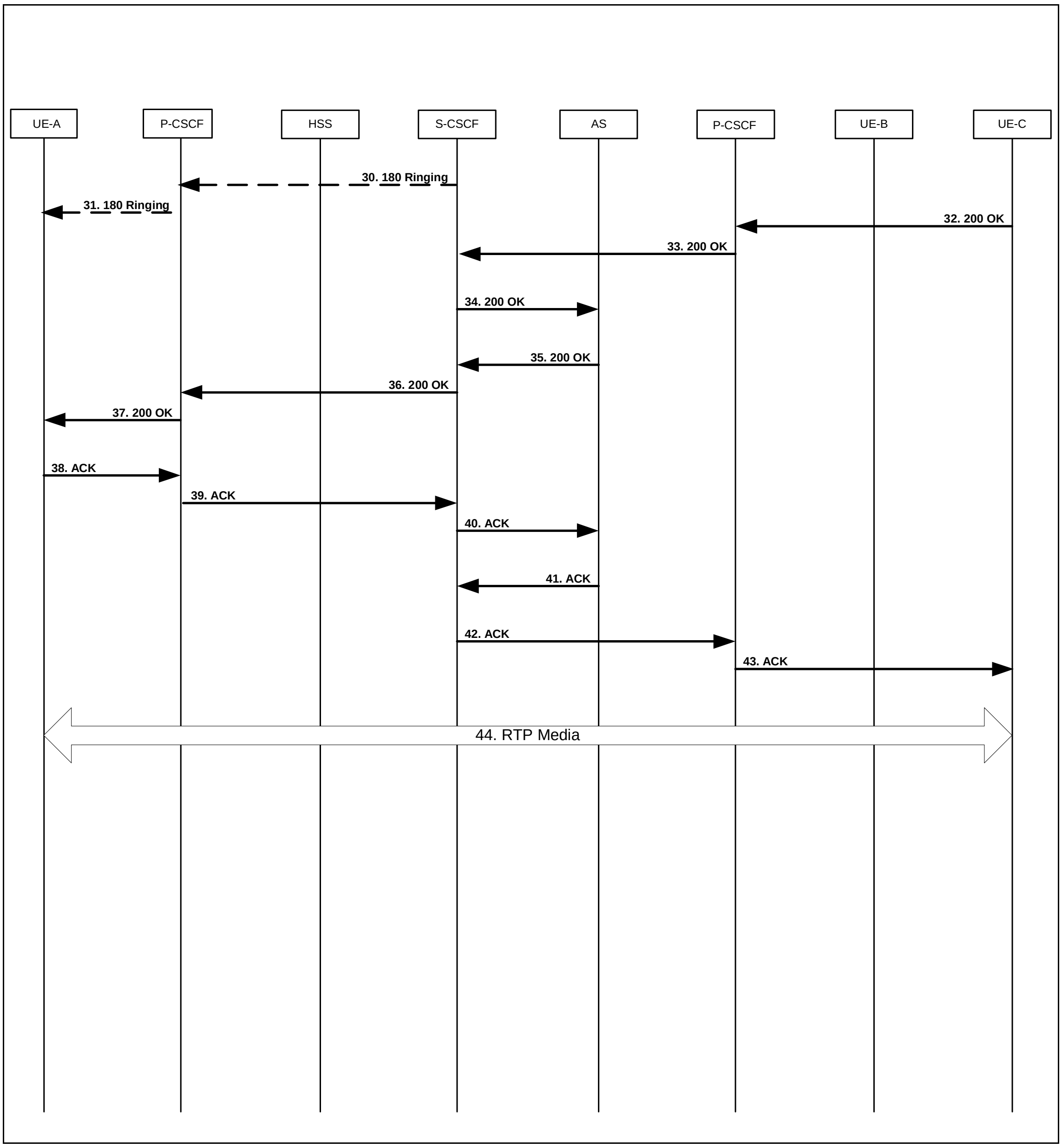
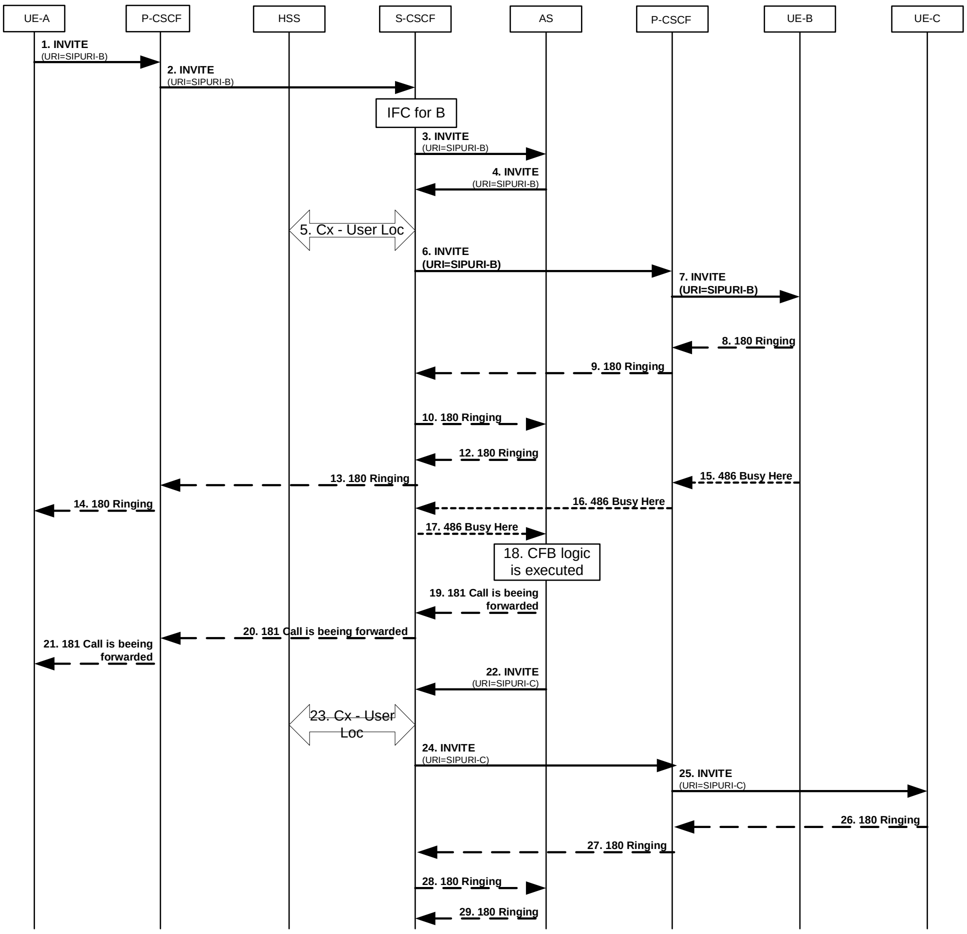

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
30. 180 Ringing
31. 180 Ringing
32. 200 OK
33. 200 OK
34. 200 OK
35. 200 OK
36. 200 OK
37. 200 OK
38. ACK
39. ACK
40. ACK
41. ACK
42. ACK
43. ACK
44. RTP Media

SSS__XXSSCFB 06	**NGN reference to: [17]**
TSS reference:	SIP-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 (user 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 b) Supported: 100 rel; Case c) Supported: 100 rel and precondition.

Comments:



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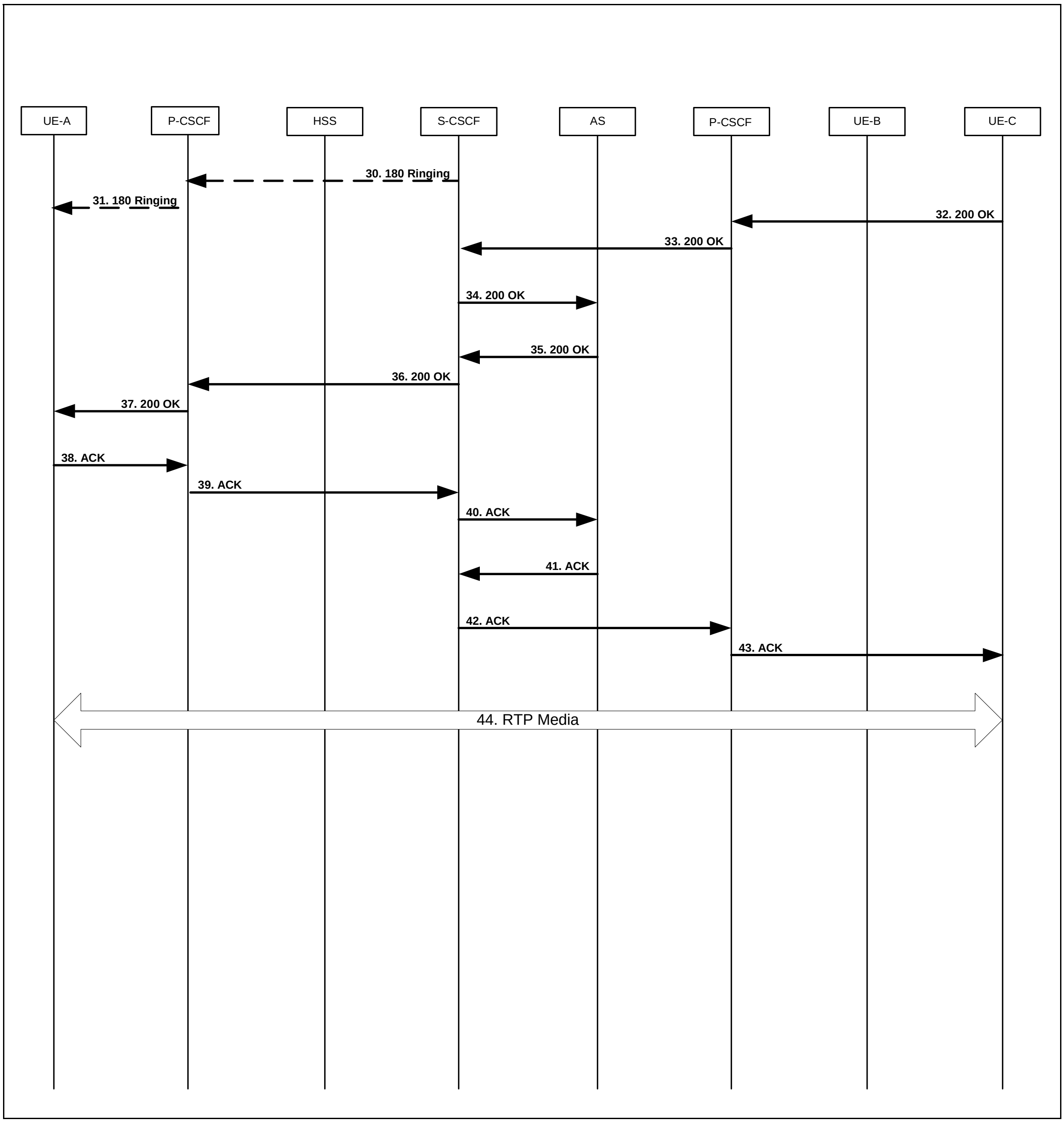
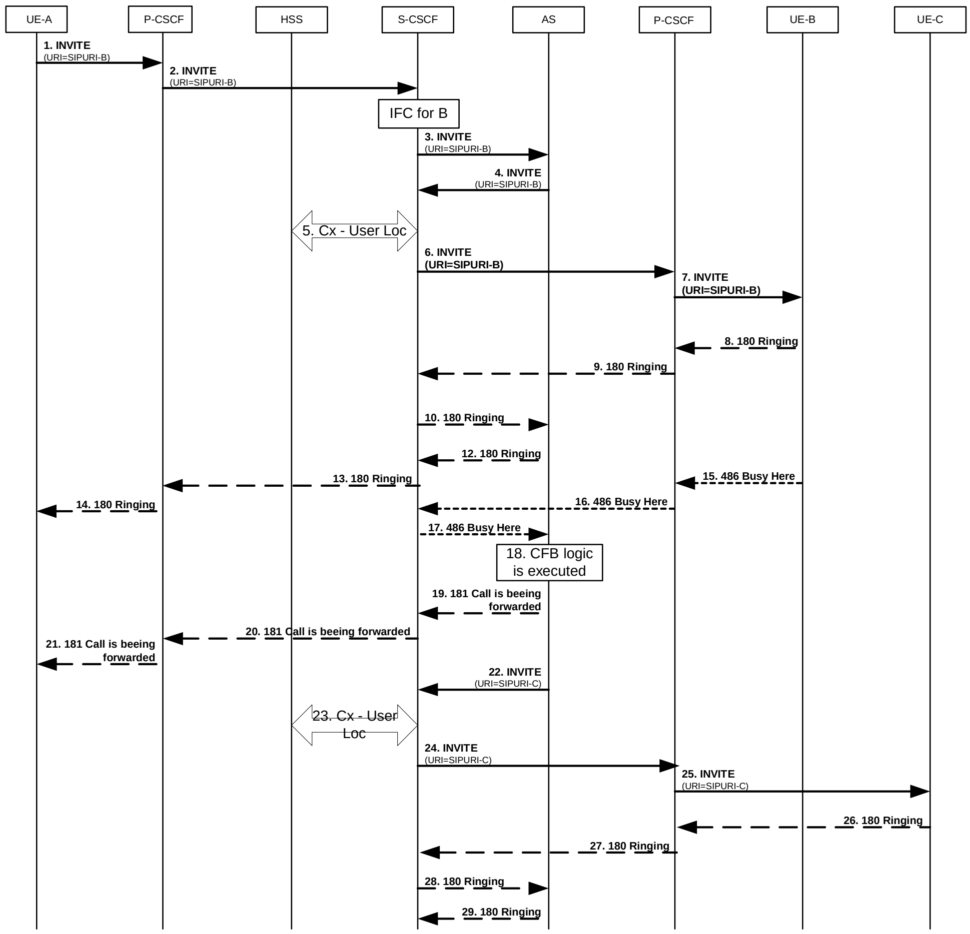

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
30. 180 Ringing
31. 180 Ringing
32. 200 OK
33. 200 OK
34. 200 OK
35. 200 OK
36. 200 OK
37. 200 OK
38. ACK
39. ACK
40. ACK
41. ACK
42. ACK
43. ACK
44. RTP Media

SSS__XXSSCFB 07	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFB
Configuration	The user A and the user C are in network N1. The user B is provided with CFB. ("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).
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 notified of call diversion and informed of the diverted-to number (user C has TIP) 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:



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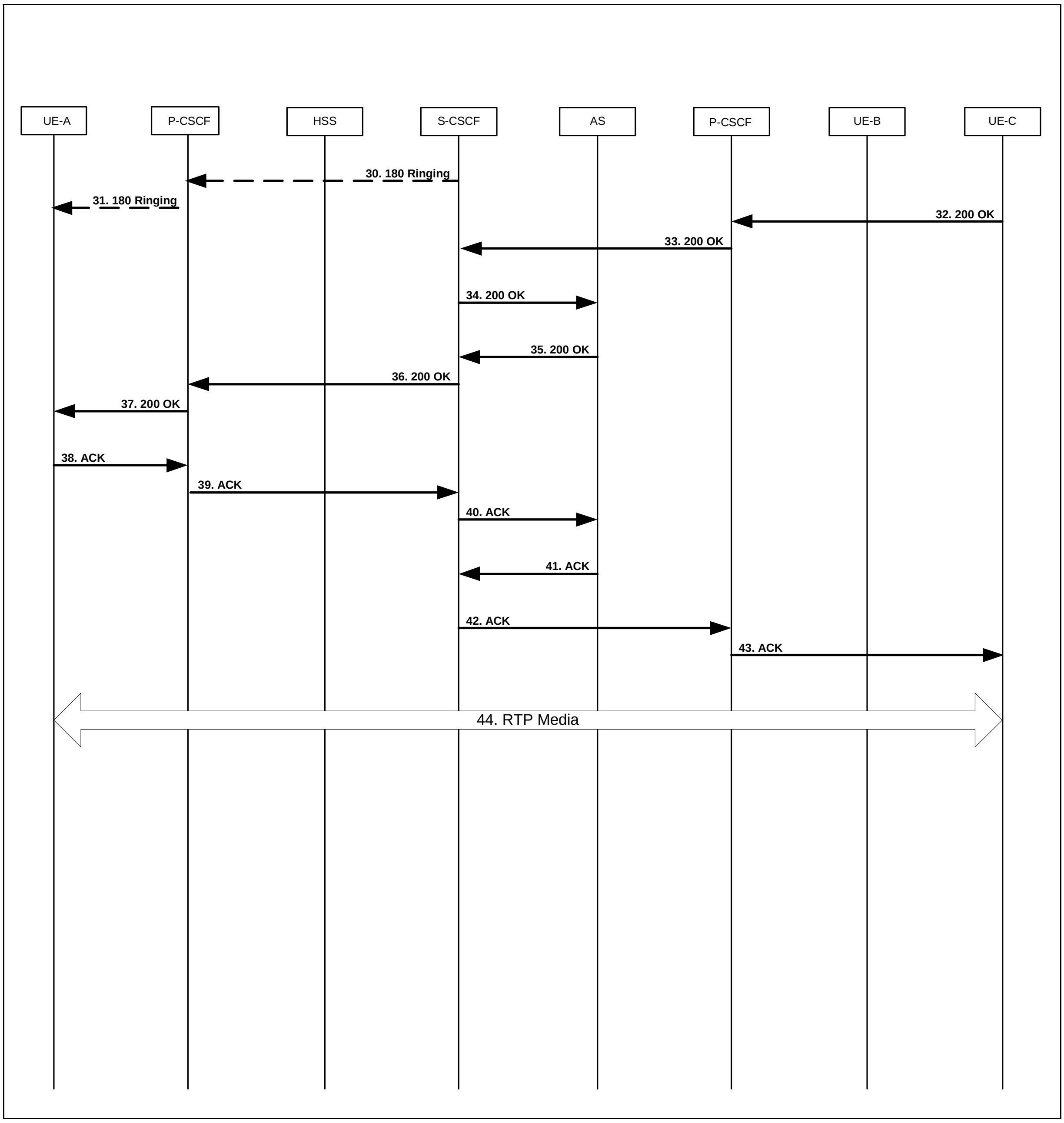
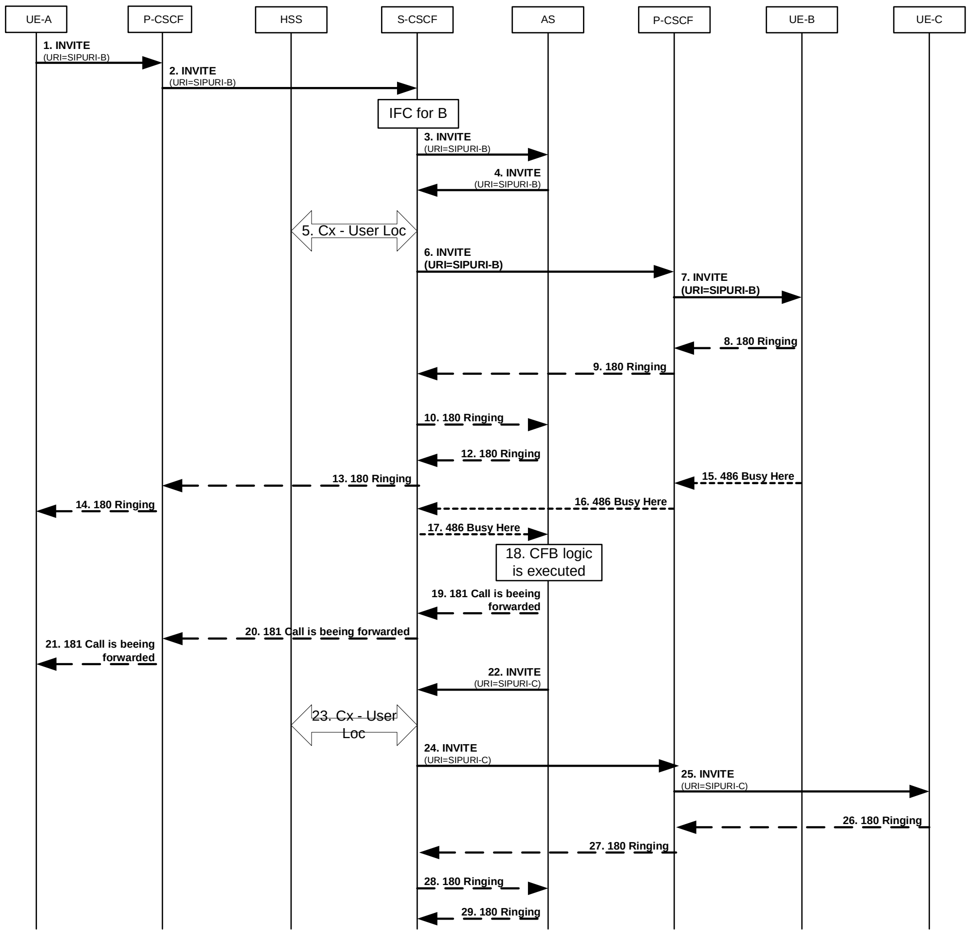

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
30. 180 Ringing
31. 180 Ringing
32. 200 OK
33. 200 OK
34. 200 OK
35. 200 OK
36. 200 OK
37. 200 OK
38. ACK
39. ACK
40. ACK
41. ACK
42. ACK
43. ACK
44. RTP Media

SSS__XXSSCFB 08	**NGN reference to: [17]**
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFB
Configuration	The user A and the user C are in network N1. The user B is provided with CFB. ("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).
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 notified of call diversion and informed of the diverted-to number (user C has TIP) 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:



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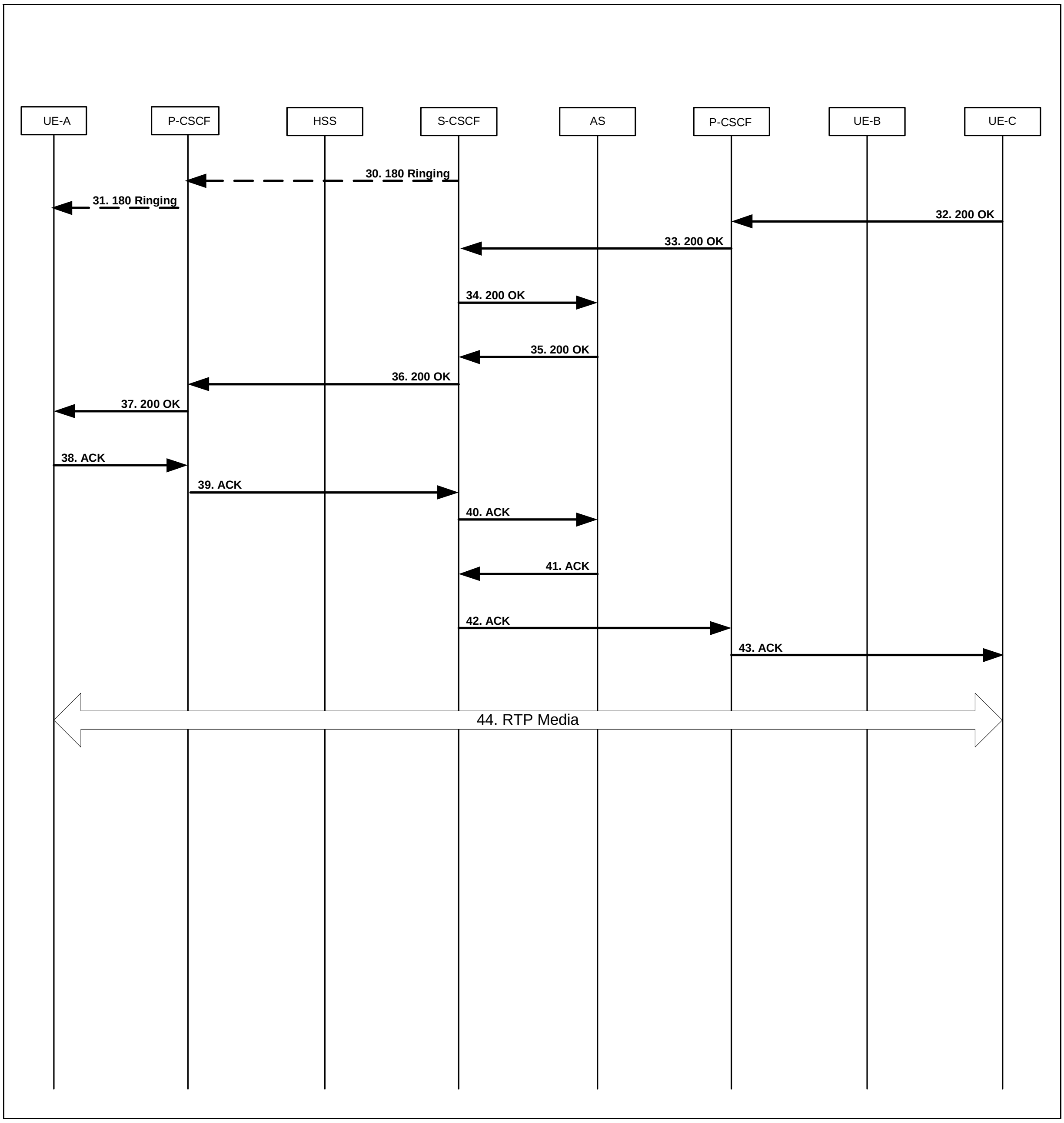
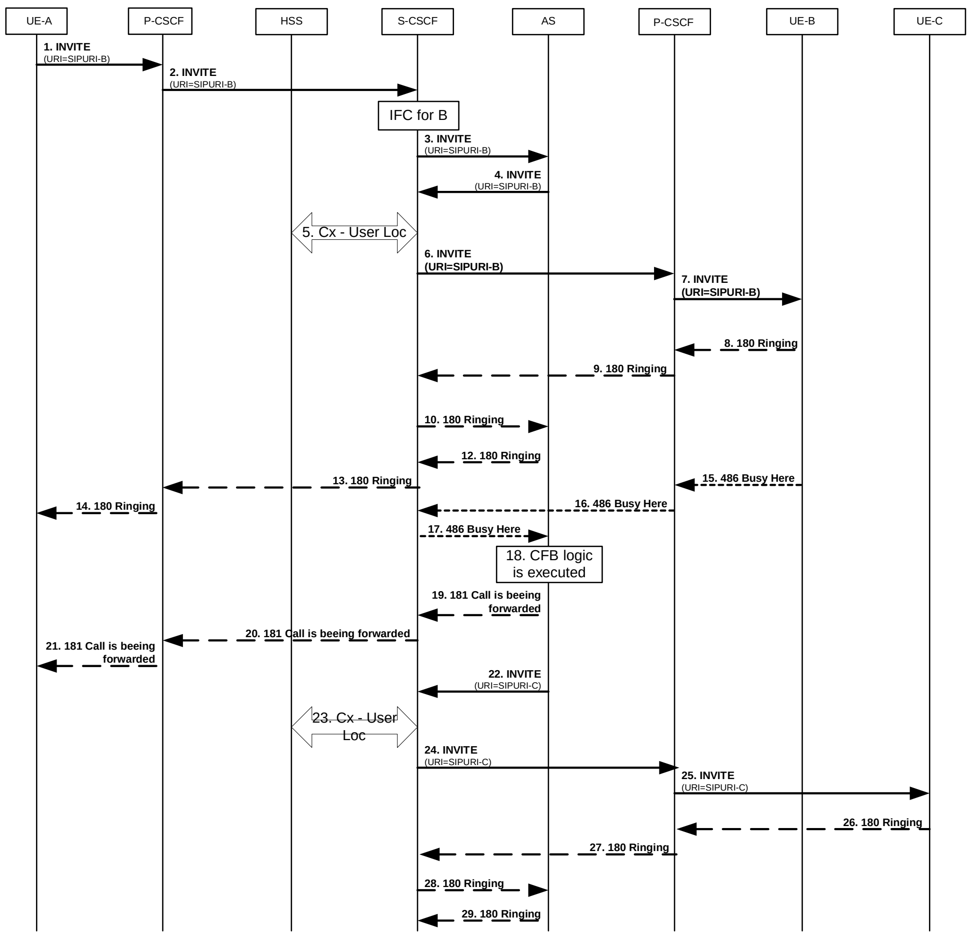

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
30. 180 Ringing
31. 180 Ringing
32. 200 OK
33. 200 OK
34. 200 OK
35. 200 OK
36. 200 OK
37. 200 OK
38. ACK
39. ACK
40. ACK
41. ACK
42. ACK
43. ACK
44. RTP Media

SSS__XXSSCFB 09	**NGN reference to: [17]**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFB	
Configuration	The user A and the user C are in network N1. 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"=Yes, "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 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:



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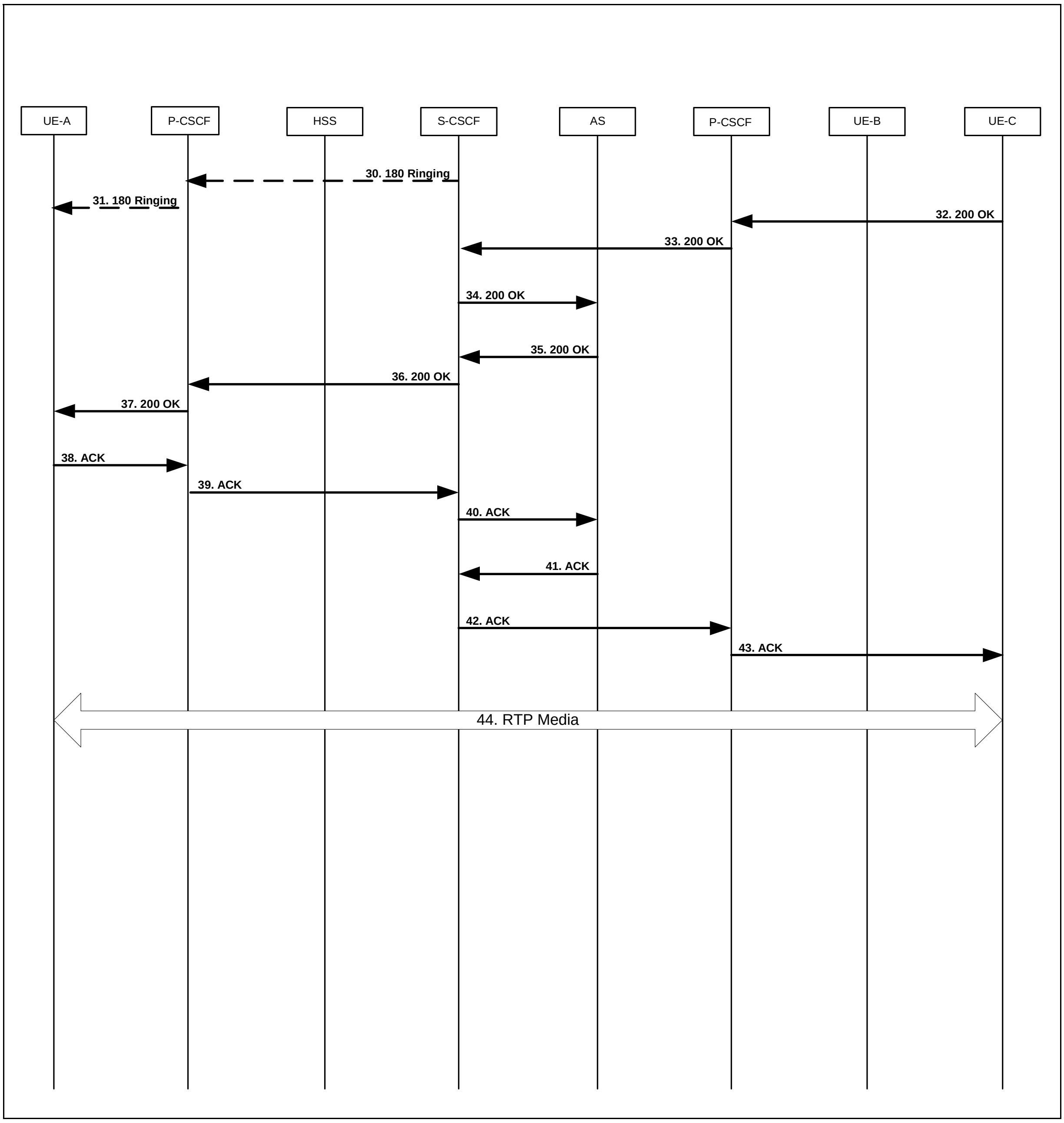
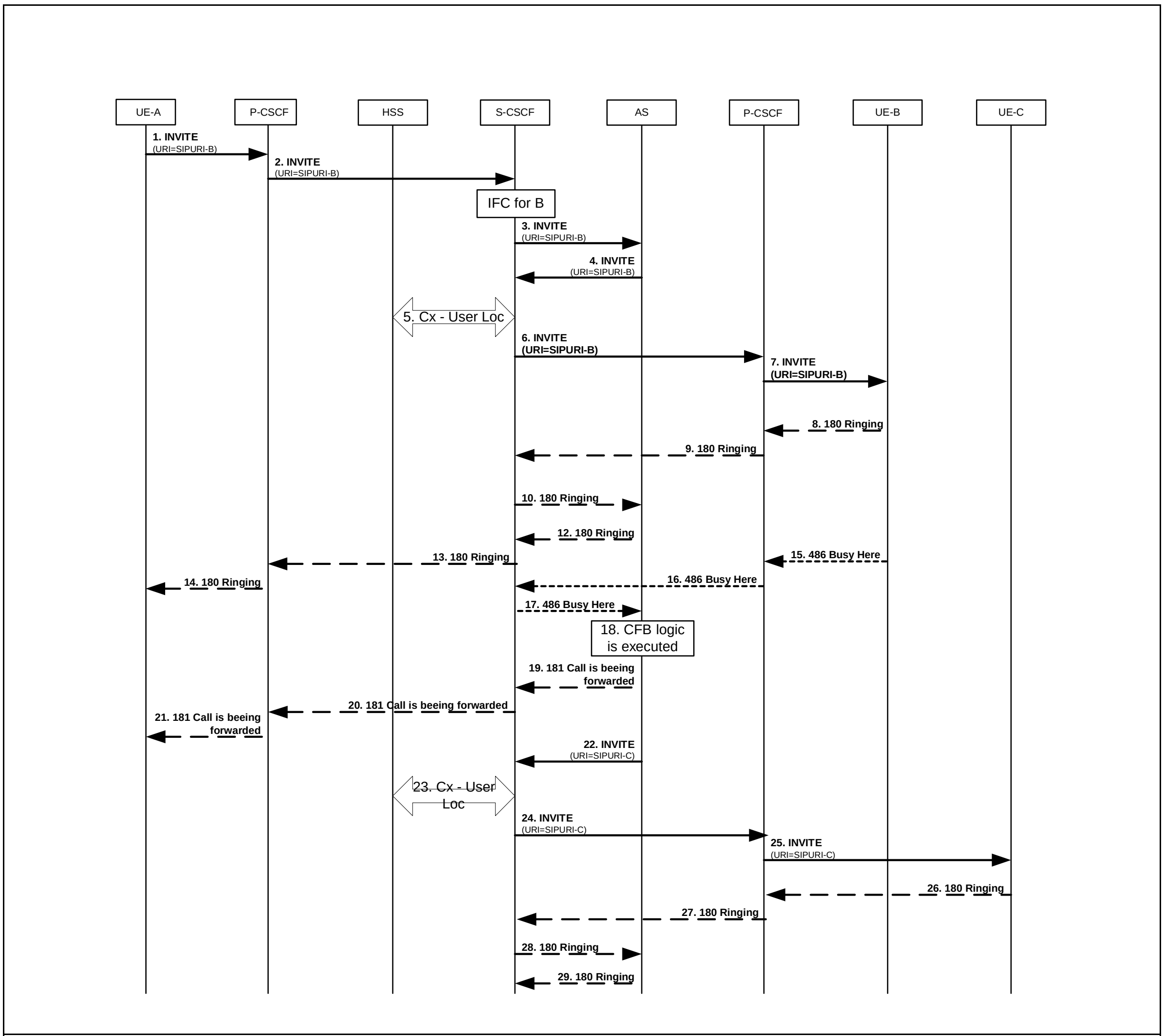
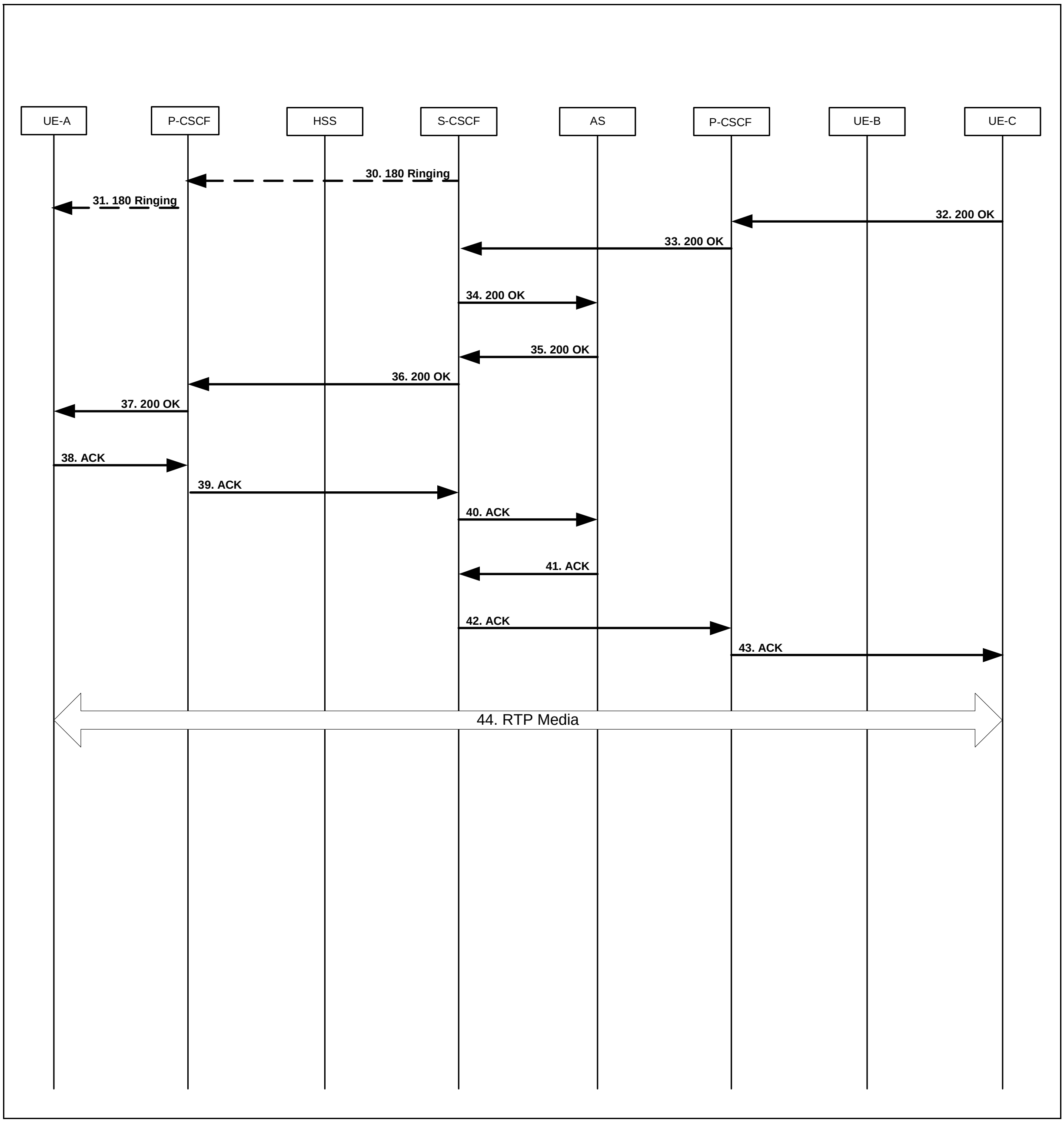

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
30. 180 Ringing
31. 180 Ringing
32. 200 OK
33. 200 OK
34. 200 OK
35. 200 OK
36. 200 OK
37. 200 OK
38. ACK
39. ACK
40. ACK
41. ACK
42. ACK
43. ACK
44. RTP Media

SSS__XXSSCFB 10	**NGN reference to: [17]**
TSS reference:	SIP-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 b) Supported: 100 rel Case c) Supported: 100 rel and precondition
Comments: Continued	


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
1. INVITE (URI=SIPURI-B)
2. INVITE (URI=SIPURI-B)
IFC for B
3. INVITE (URI=SIPURI-B)
4. INVITE (URI=SIPURI-B)
5. Cx - User Loc
6. INVITE (URI=SIPURI-B)
7. INVITE (URI=SIPURI-B)
8. 180 Ringing
9. 180 Ringing
10. 180 Ringing
12. 180 Ringing
13. 180 Ringing
14. 180 Ringing
15. 486 Busy Here
16. 486 Busy Here
17. 486 Busy Here
18. CFB logic is executed
19. 181 Call is beeing forwarded
20. 181 Call is beeing forwarded
21. 181 Call is beeing forwarded
22. INVITE (URI=SIPURI-C)
23. Cx - User Loc
24. INVITE (URI=SIPURI-C)
25. INVITE (URI=SIPURI-C)
26. 180 Ringing
27. 180 Ringing
28. 180 Ringing
29. 180 Ringing

Continued



SSS__XXSSCFB 15	**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 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:	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 user determined user busy.		
Test purpose:	The served user subscribes to the CFB simulation service; originating user is notified. Additionaly an announcement is played. When Communication Diversion occurs and if the notification procedures of the originating user is supported then a 181 (Call Is Being Forwarded) response shall be sent towards the originating user. Additional the AS initiates an announcement to be included towards the calling user in order to inform about the diversion. Announcements shall be played according to procedures as are described in TS 183 028 [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 **SIP header values:** 181 Call is Being Forwarded		
Comments:			

SIP#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE				
		INVITE	→	INVITE		
		486 Busy Here	←	486 Busy Here		
		ACK	→	ACK		
181 Call is Being Forwarded	←	181 Call is Being Forwarded				
		INVITE	→		→	INVITE
		180 Ringing	←		←	180 Ringing
180 Ringing	←	180 Ringing				
		200 OK (INVITE)	←		←	200 OK (INVITE)
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)
The test case can not be tested with an ATS						

SSS__XXSSCFB 18	**Reference TS 183 004 [17] clause 4.5.2.6.3**	
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 user determined user busy.	
Test purpose:	The served user subscribes to the CFB simulation service; Indication of communication diversion to the diverting user using the MESSAGE request triggered by a timer. Ensure that when the diverting user is registering the AS sends a MESSAGE request including the information where the call is diverted to. The MESSAGE request that is be sent due to an timer value that can be provided by the 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#1		SUT		SIP#2 (served user)		SIP#3
INVITE 1	→	INVITE				
		INVITE	→	INVITE		
		486 Busy Here	←	486 Busy Here		
		ACK	→	ACK		
		INVITE 2	→		→	INVITE
		MESSAGE	→	MESSAGE		
		200 OK	←	200 OK		
		180 Ringing	←		←	180 Ringing
180 Ringing	←	180 Ringing				
		200 OK (INVITE)	←		←	200 OK (INVITE)
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)

SSS__XXSSCFB 19	**Reference TS 183 004 [17] clause 4.5.2.6.3**	
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 user determined user busy.	
Test purpose:	The served user subscribes to the CFB simulation service; Indication of communication diversion to the diverting user using the MESSAGE request when a new outgoing communication is requested. Ensure that a diverting user will be informed with a MESSAGE request after the diverting user has initiated a new outgoing communication the information where the 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 Case c) Supported: 100 rel and precondition	

Comments:

SIP#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE				
		INVITE	→	INVITE		
		486 Busy Here	←	486 Busy Here		
		ACK	→	ACK		
		INVITE 2	→		→	INVITE
		MESSAGE	→	MESSAGE		
		180 Ringing	←		←	180 Ringing
		200 OK	←	200 OK		
180 Ringing	←	180 Ringing				
		200 OK (INVITE)	←		←	200 OK (INVITE)
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→				→	ACK
		Communication				
		INVITE 2	→		→	INVITE
		MESSAGE	→	MESSAGE		
		180 Ringing	←		←	180 Ringing
			←	200 OK (INVITE)	←	200 OK (INVITE)
			→	ACK	→	ACK

The test case can not 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/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:	The served user subscribes to the CFB simulation service; Indication of communication diversion to the diverting user using a voicemail or mail system. Ensure that when communication diversion is performed, the AS sends Voicemail or Message mail system request including the information where the 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 Case c) Supported: 100 rel and precondition	

Comments:

SIP#1		**SUT**			**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE					
			INVITE	→	INVITE		
			486 Busy Here	←	486 Busy Here		
			ACK	→	ACK		
			INVITE 2	→		→	INVITE
			Voicemail or Message mail system	→	**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	→					→	ACK
			Communication				
BYE	→					→	BYE
200 OK (BYE)	←					←	200 OK (BYE)

The test case can not be tested with an ATS

SSS__XXSSCFB 21	**Reference** **TS 183 004 [17] clause 4.5.2.6.3**	
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 user determined user busy.	
Test purpose:	The served user subscribes to the CFB simulation service; Indication of communication diversion to the diverting user using a voicemail or mail system requested by a timer. Ensure that when communication diversion is performed, the AS sends a Voicemail or Message mail system request including the information where the call is diverted to. The Voicemail or Message mail system shall be sent due to a timer value that can be provided by the 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#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE				
		INVITE	→	INVITE		
		486 Busy Here	←	486 Busy Here		
		ACK	→	ACK		
		INVITE 2	→		→	INVITE
		Voicemail or Message mail system	→	**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	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)

The test case can not be tested with an ATS

SSS__XXSSCFB 22	**Reference TS 183 004 [17] clause 4.5.2.6.3**	
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 user determined user busy.	
Test purpose:	The served user subscribes to the CFB simulation service; periodically indication of communication diversion to the diverting user using a voicemail or mail system. Ensure that when communication diversion is performed. the AS sends a Voicemail or Message mail system request including the information where the call is diverted to. The diverting user will be informed periodically with Voicemail or Message mail system with the information where the 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 Case c) Supported: 100 rel and precondition	

Comments:

SIP#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE				
		INVITE	→	INVITE		
		486 Busy Here	←	486 Busy Here		
		ACK	→	ACK		
		INVITE 2	→		→	INVITE
		Voicemail or Message mail system	→	**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	→				→	ACK
		Voicemail or Message mail system	→	**Voicemail or Message mail system**		
		Communication				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)

The test case can not be tested with an ATS

SSS__XXSSCFB 23	**Reference TS 183 004 [17] clause 4.5.2.6.3**	
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 user determined user busy.	
Test purpose:	The served user subscribes to the CFB simulation service; Indication of a new communication diversion to the diverting user using a voicemail or mail system. Ensure that a diverting user will be informed with Message mail system request after the diverting user has initiated a new outgoing communication the information where the 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 Case c) Supported: 100 rel and precondition	

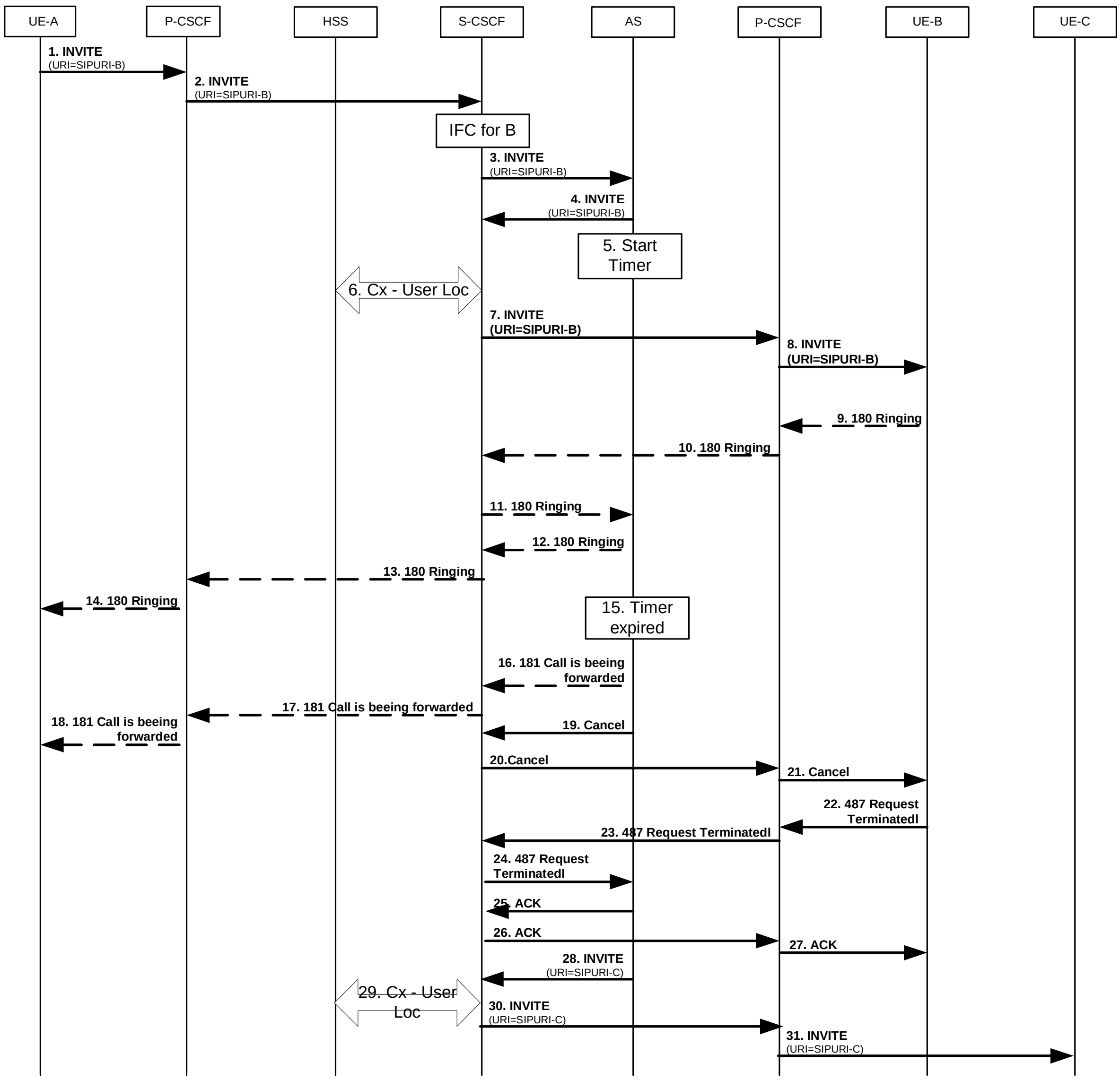
Comments:

SIP#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE				
		INVITE	→	INVITE		
		486 Busy Here	←	486 Busy Here		
		ACK	→	ACK		
		Voicemail or Message mail system	→	**Voicemail or 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
		Communication				
		INVITE 3	→		→	INVITE
		180 Ringing	←		←	180 Ringing
			←	180 Ringing		
			←	200 OK (INVITE)	←	200 OK (INVITE)
		Voicemail or Message mail system	→	**Voicemail or Message mail system**		
			→	ACK	→	ACK

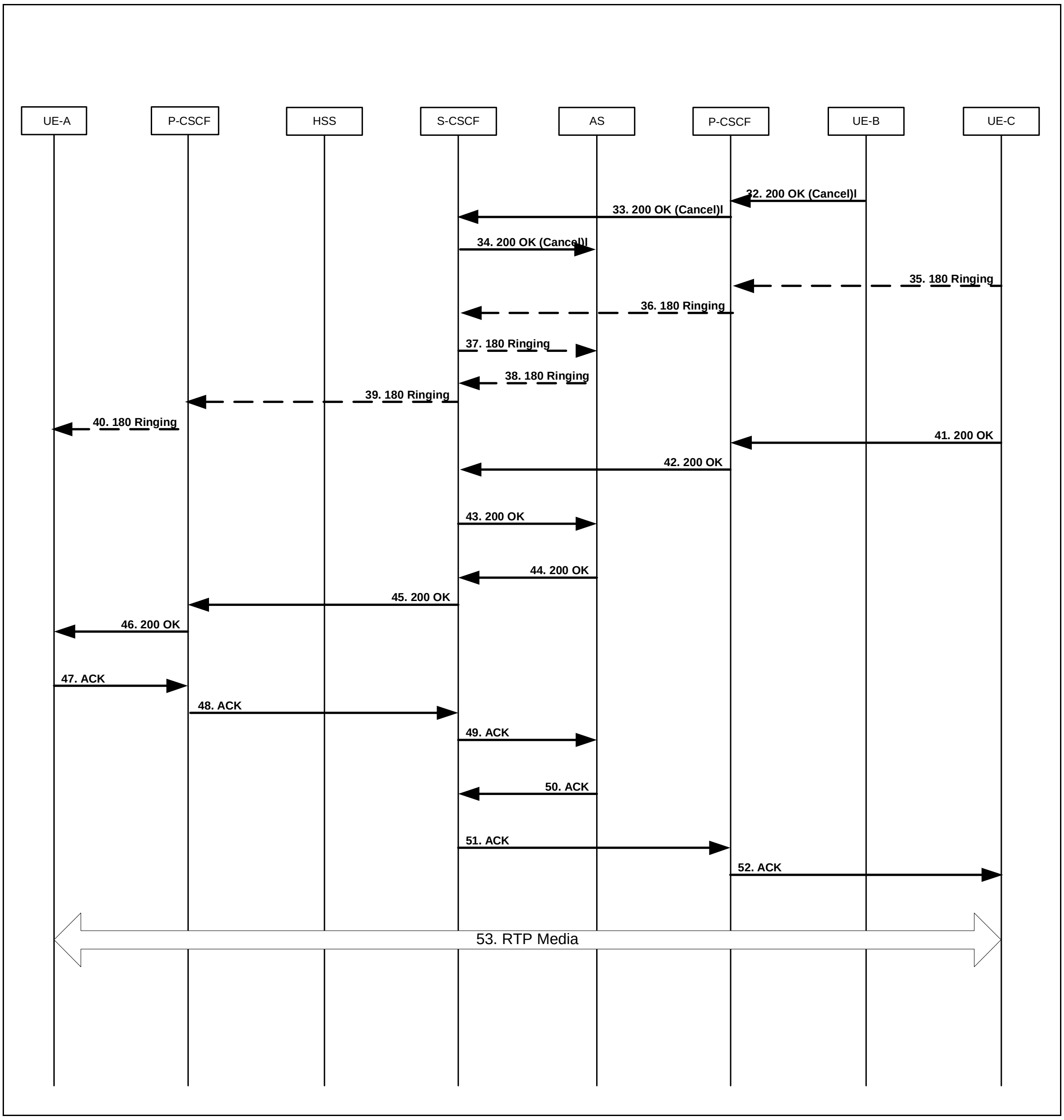
The test case can not be tested with an ATS

6.5.2.8 CFNR

SSS__XXSSCFNR01	NGN reference to: [17]
TSS reference:	SIP-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].
Test 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:	

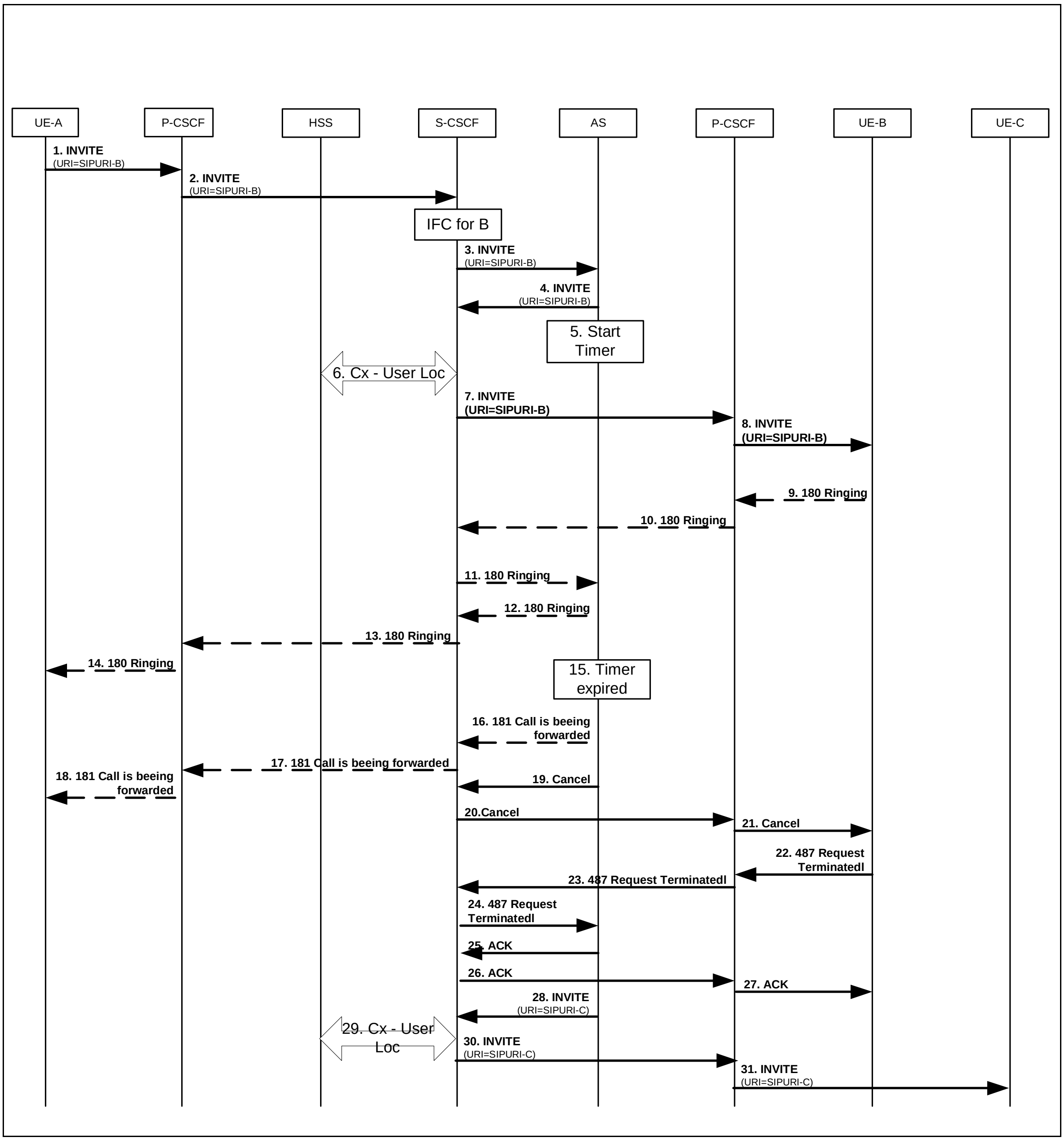


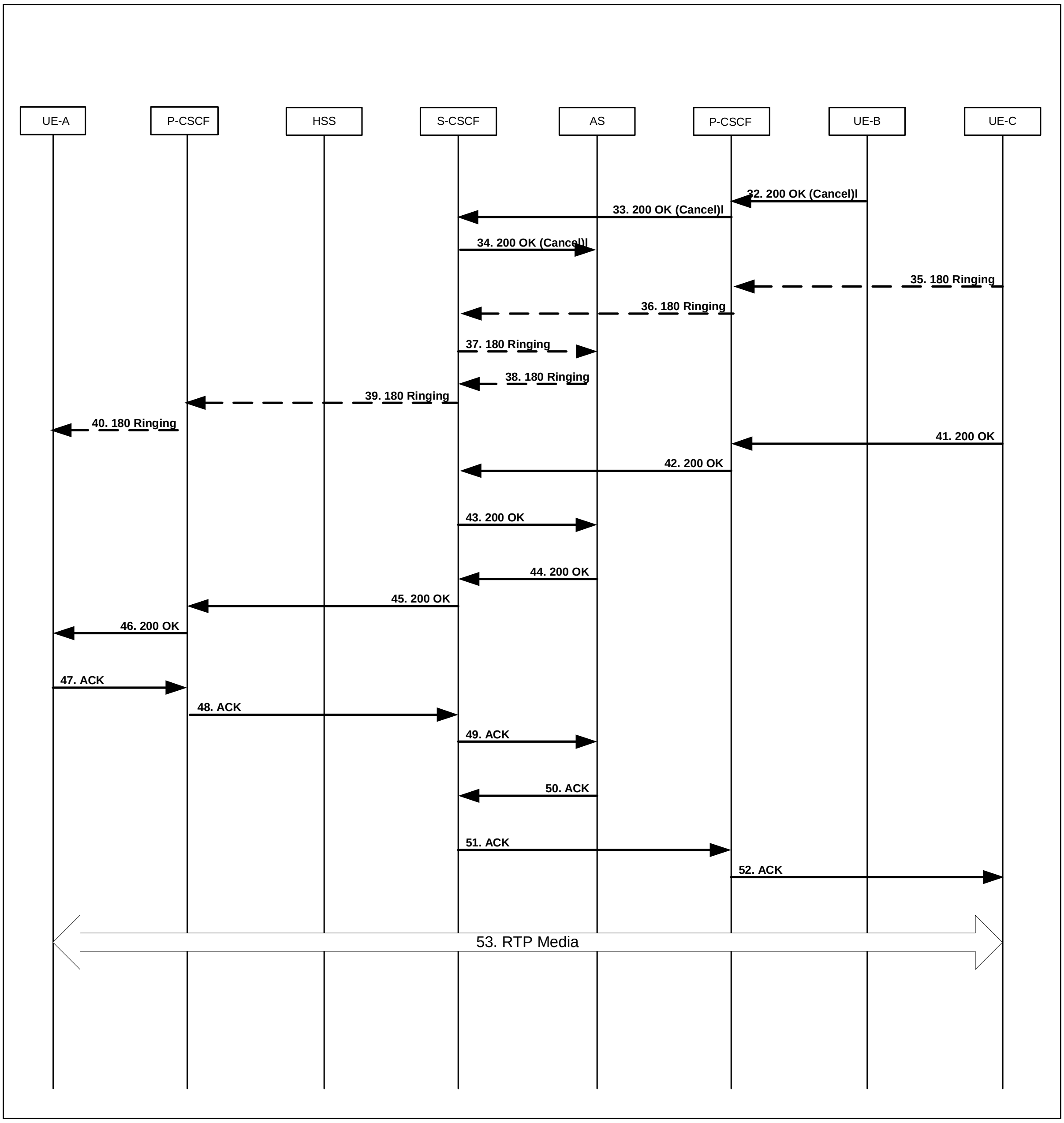
Continued


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
32. 200 OK (Cancel)l
33. 200 OK (Cancel)l
34. 200 OK (Cancel)l
35. 180 Ringing
36. 180 Ringing
37. 180 Ringing
38. 180 Ringing
39. 180 Ringing
40. 180 Ringing
41. 200 OK
42. 200 OK
43. 200 OK
44. 200 OK
45. 200 OK
46. 200 OK
47. ACK
48. ACK
49. ACK
50. ACK
51. ACK
52. ACK
53. RTP Media

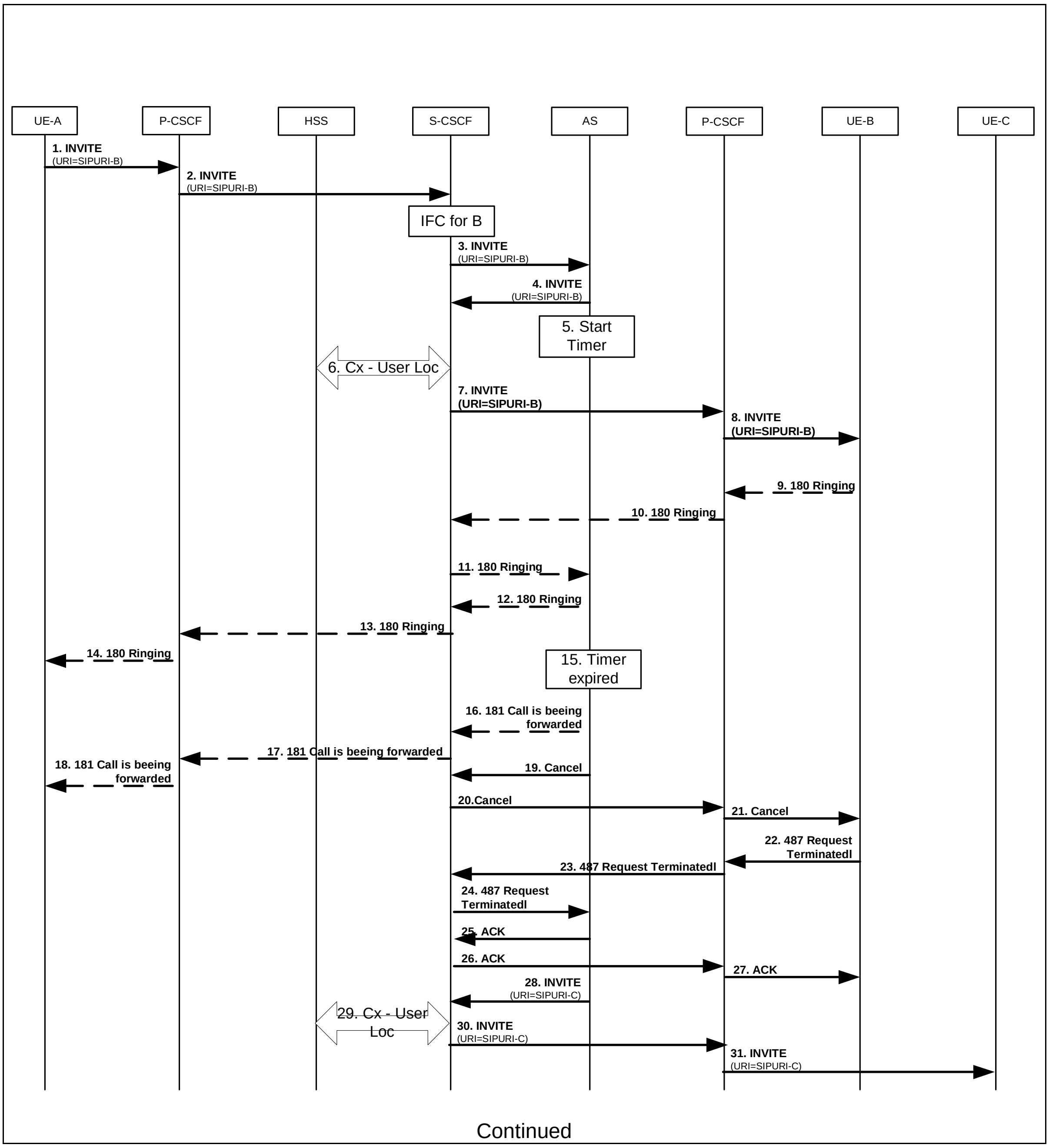
SSS__XXSSCFNR02	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNR/
Configuration	The user A and the user C are in network N1. The user B is provided with CFNR. ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting 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: o 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:	

Continued

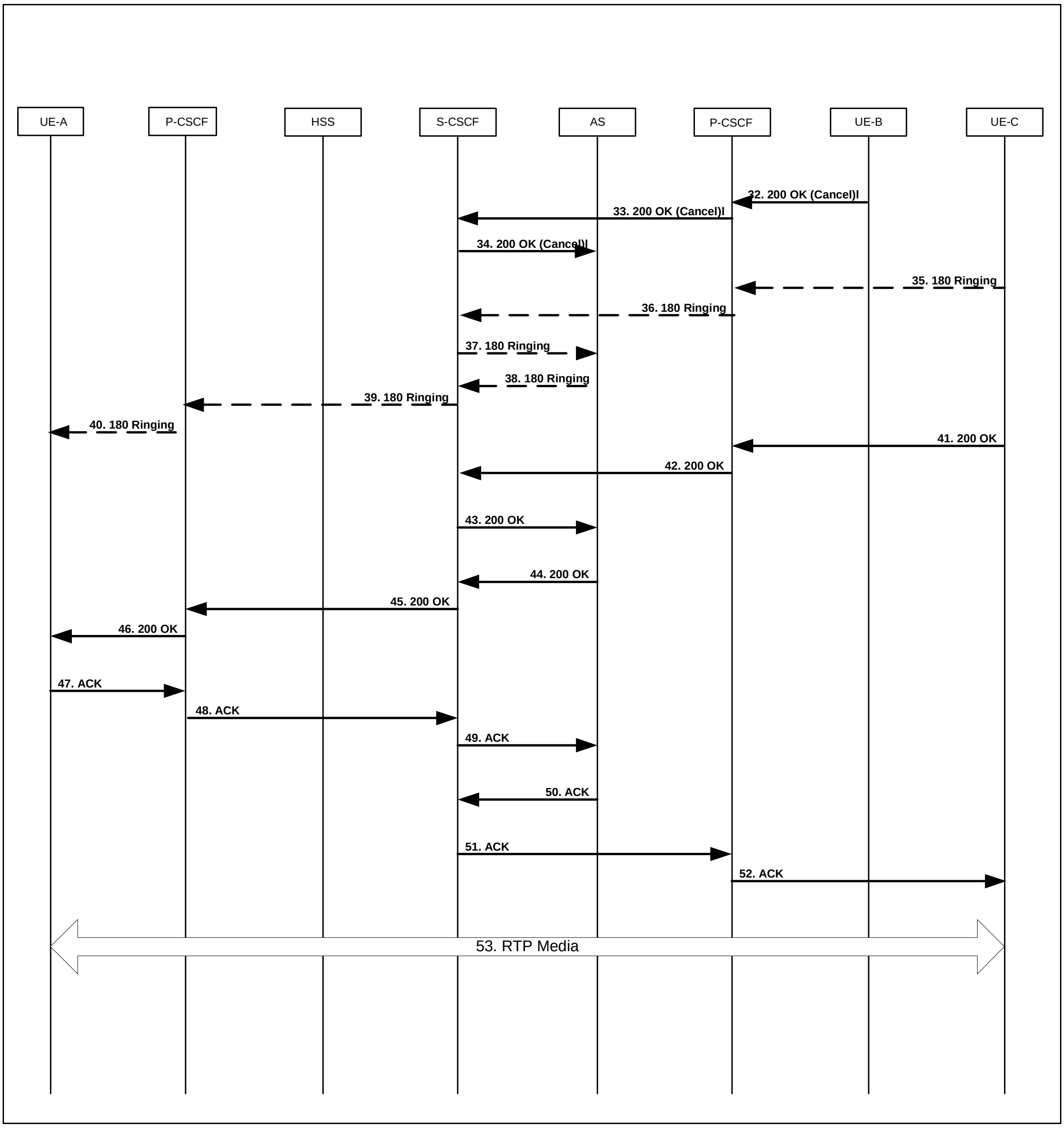

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
1. INVITE (URI=SIPURI-B)
2. INVITE (URI=SIPURI-B)
IFC for B
3. INVITE (URI=SIPURI-B)
4. INVITE (URI=SIPURI-B)
5. Start Timer
6. Cx - User Loc
7. INVITE (URI=SIPURI-B)
8. INVITE (URI=SIPURI-B)
9. 180 Ringing
10. 180 Ringing
11. 180 Ringing
12. 180 Ringing
13. 180 Ringing
14. 180 Ringing
15. Timer expired
16. 181 Call is beeing forwarded
17. 181 Call is beeing forwarded
18. 181 Call is beeing forwarded
19. Cancel
20.Cancel
21. Cancel
22. 487 Request Terminatedl
23. 487 Request Terminatedl
24. 487 Request Terminatedl
25. ACK
26. ACK
27. ACK
28. INVITE (URI=SIPURI-C)
29. Cx - User Loc
30. INVITE (URI=SIPURI-C)
31. INVITE (URI=SIPURI-C)


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
32. 200 OK (Cancel)l
33. 200 OK (Cancel)l
34. 200 OK (Cancel)l
35. 180 Ringing
36. 180 Ringing
37. 180 Ringing
38. 180 Ringing
39. 180 Ringing
40. 180 Ringing
41. 200 OK
42. 200 OK
43. 200 OK
44. 200 OK
45. 200 OK
46. 200 OK
47. ACK
48. ACK
49. ACK
50. ACK
51. ACK
52. ACK
53. RTP Media

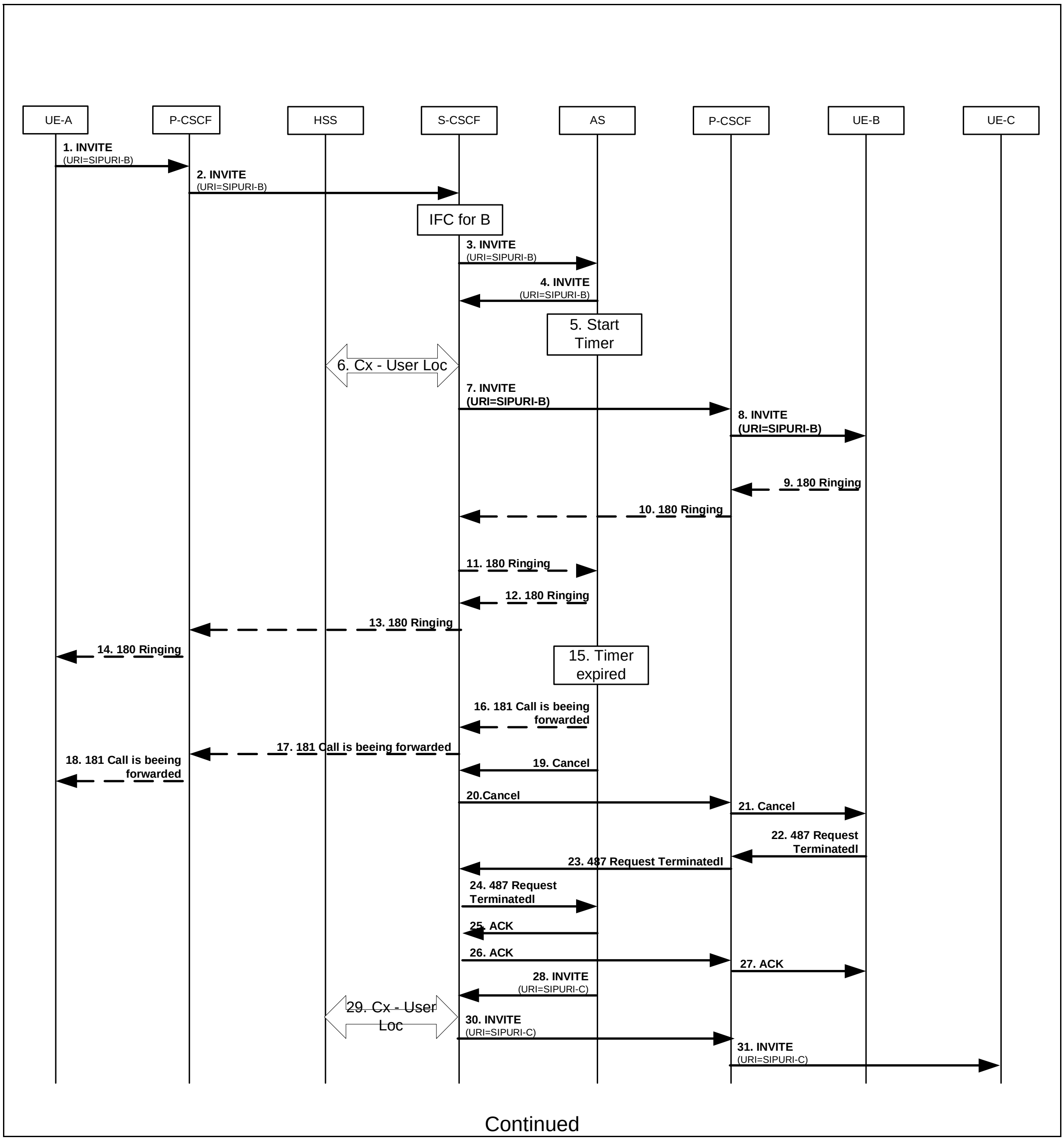
SSS__XXSSCFNR03	NGN reference to: [17]	
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNR/	
Configuration	The user A and the user C are in network N1. The user B is provided with CFNR. ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting 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: o 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:		

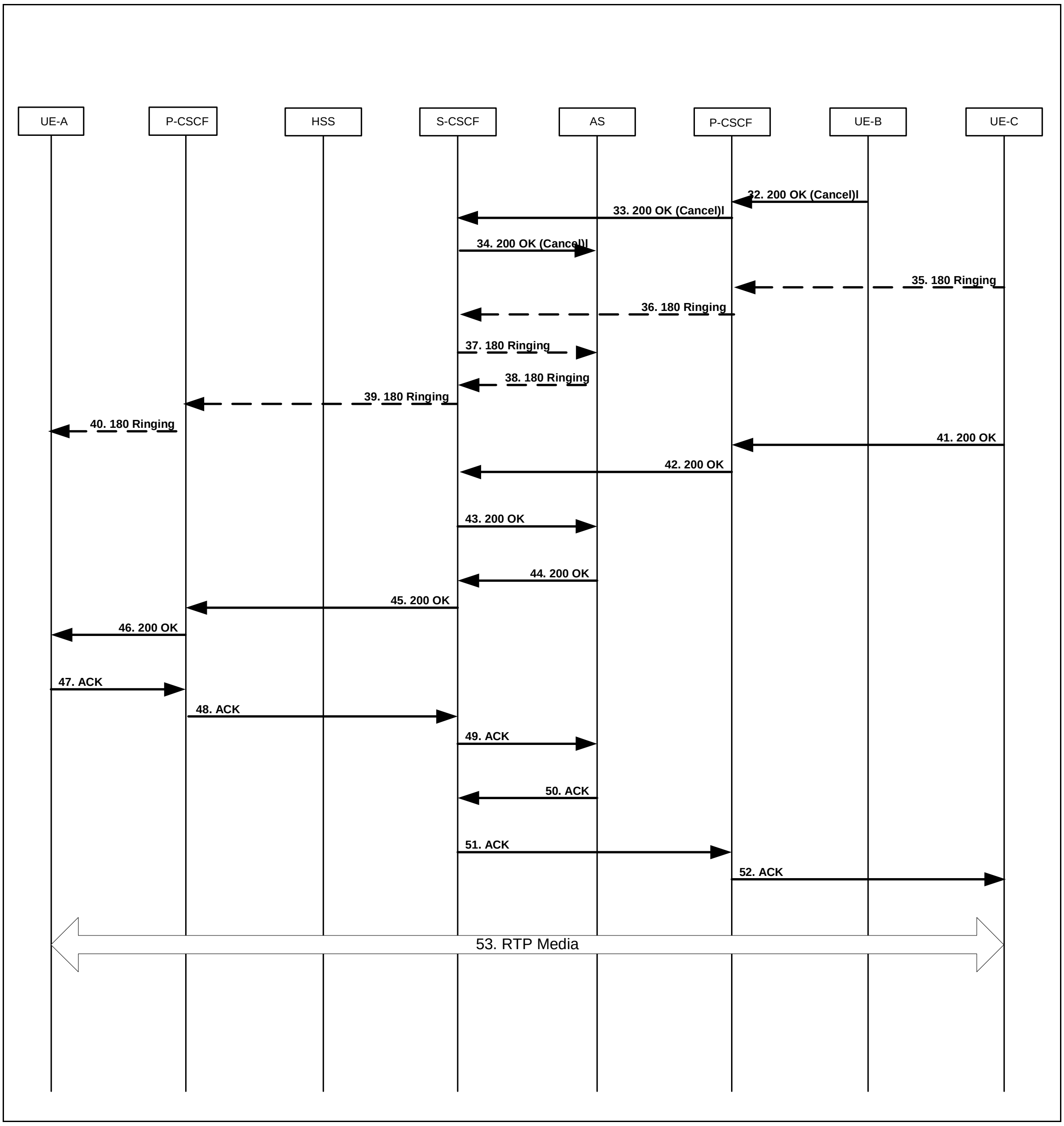


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
1. INVITE
(URI=SIPURI-B)
2. INVITE
(URI=SIPURI-B)
IFC for B
3. INVITE
(URI=SIPURI-B)
4. INVITE
(URI=SIPURI-B)
5. Start Timer
6. Cx - User Loc
7. INVITE
(URI=SIPURI-B)
8. INVITE
(URI=SIPURI-B)
9. 180 Ringing
10. 180 Ringing
11. 180 Ringing
12. 180 Ringing
13. 180 Ringing
14. 180 Ringing
15. Timer expired
16. 181 Call is beeing forwarded
17. 181 Call is beeing forwarded
18. 181 Call is beeing forwarded
19. Cancel
20.Cancel
21. Cancel
22. 487 Request Terminatedl
23. 487 Request Terminatedl
24. 487 Request Terminatedl
25. ACK
26. ACK
27. ACK
28. INVITE
(URI=SIPURI-C)
29. Cx - User Loc
30. INVITE
(URI=SIPURI-C)
31. INVITE
(URI=SIPURI-C)
Continued

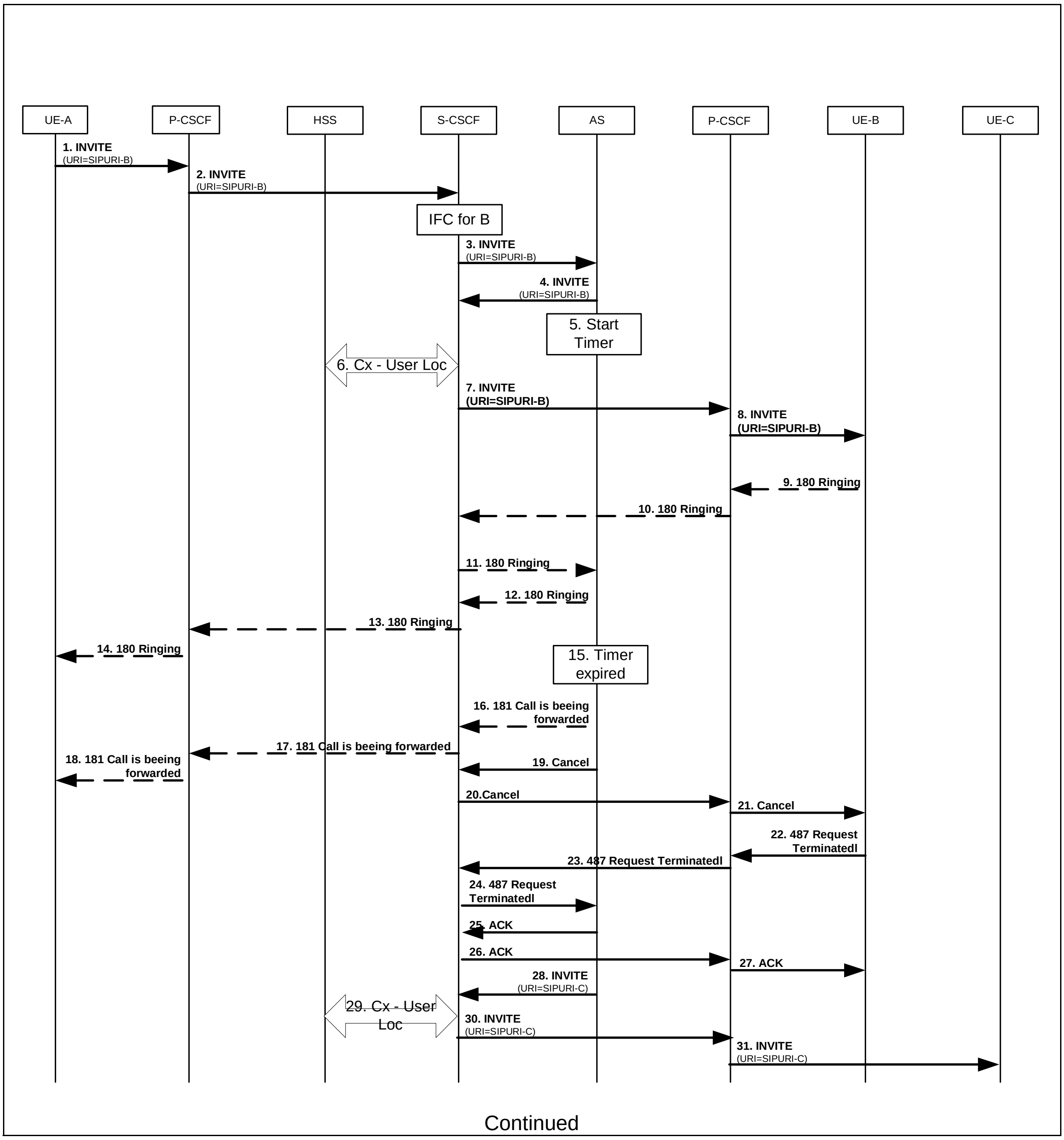

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
32. 200 OK (Cancel)l
33. 200 OK (Cancel)l
34. 200 OK (Cancel)l
35. 180 Ringing
36. 180 Ringing
37. 180 Ringing
38. 180 Ringing
39. 180 Ringing
40. 180 Ringing
41. 200 OK
42. 200 OK
43. 200 OK
44. 200 OK
45. 200 OK
46. 200 OK
47. ACK
48. ACK
49. ACK
50. ACK
51. ACK
52. ACK
53. RTP Media

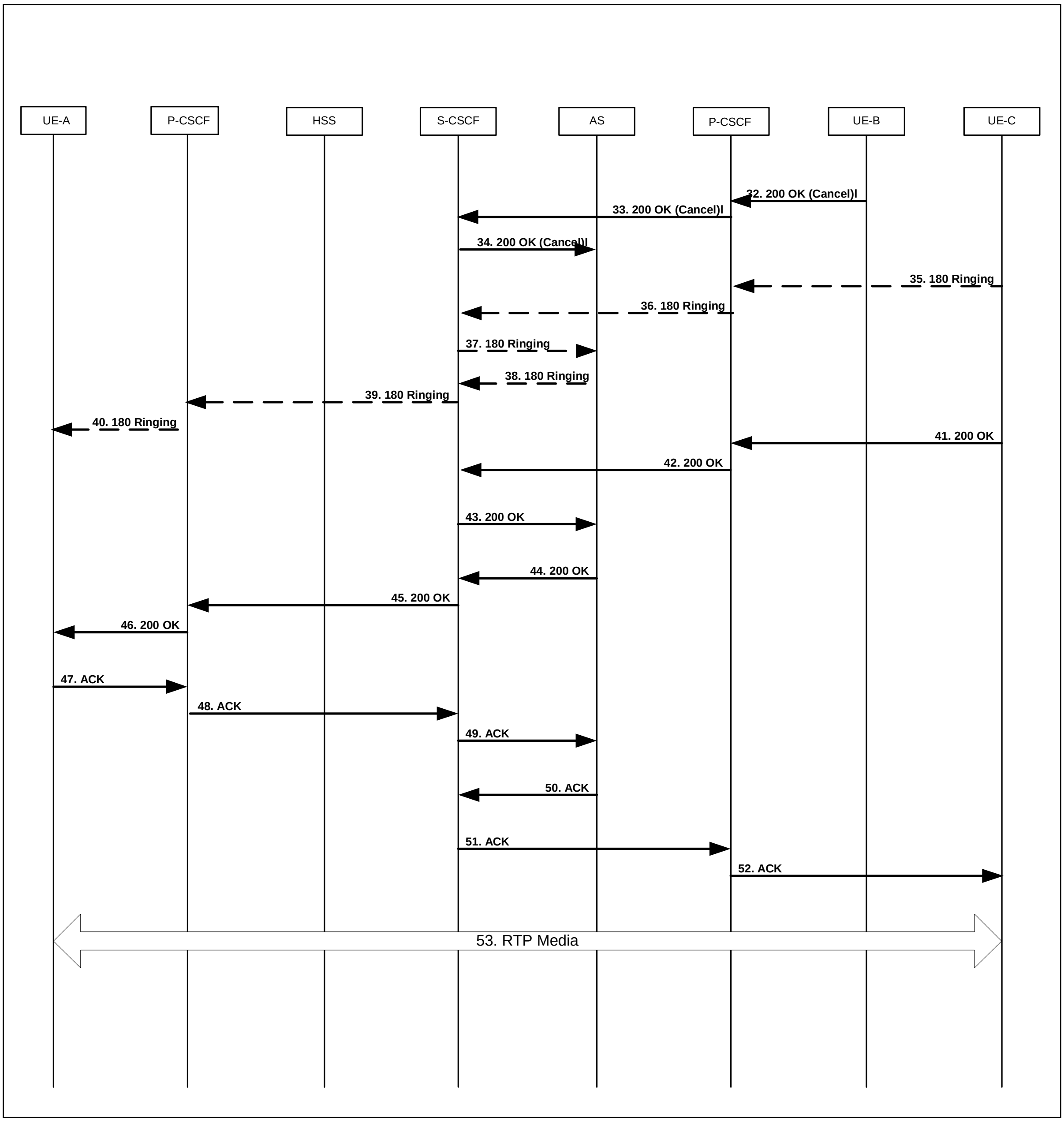
SSS__XXSSCFNR 04	**NGN reference to: [17]**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNR	
Configuration	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNR ("Served user allows the presentation of forwarded to URI to originating user in diversion 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:		


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
1. INVITE (URI=SIPURI-B)
2. INVITE (URI=SIPURI-B)
IFC for B
3. INVITE (URI=SIPURI-B)
4. INVITE (URI=SIPURI-B)
5. Start Timer
6. Cx - User Loc
7. INVITE (URI=SIPURI-B)
8. INVITE (URI=SIPURI-B)
9. 180 Ringing
10. 180 Ringing
11. 180 Ringing
12. 180 Ringing
13. 180 Ringing
14. 180 Ringing
15. Timer expired
16. 181 Call is beeing forwarded
17. 181 Call is beeing forwarded
18. 181 Call is beeing forwarded
19. Cancel
20.Cancel
21. Cancel
22. 487 Request Terminatedl
23. 487 Request Terminatedl
24. 487 Request Terminatedl
25. ACK
26. ACK
27. ACK
28. INVITE (URI=SIPURI-C)
29. Cx - User Loc
30. INVITE (URI=SIPURI-C)
31. INVITE (URI=SIPURI-C)
Continued


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
32. 200 OK (Cancel)l
33. 200 OK (Cancel)l
34. 200 OK (Cancel)l
35. 180 Ringing
36. 180 Ringing
37. 180 Ringing
38. 180 Ringing
39. 180 Ringing
40. 180 Ringing
41. 200 OK
42. 200 OK
43. 200 OK
44. 200 OK
45. 200 OK
46. 200 OK
47. ACK
48. ACK
49. ACK
50. ACK
51. ACK
52. ACK
53. RTP Media

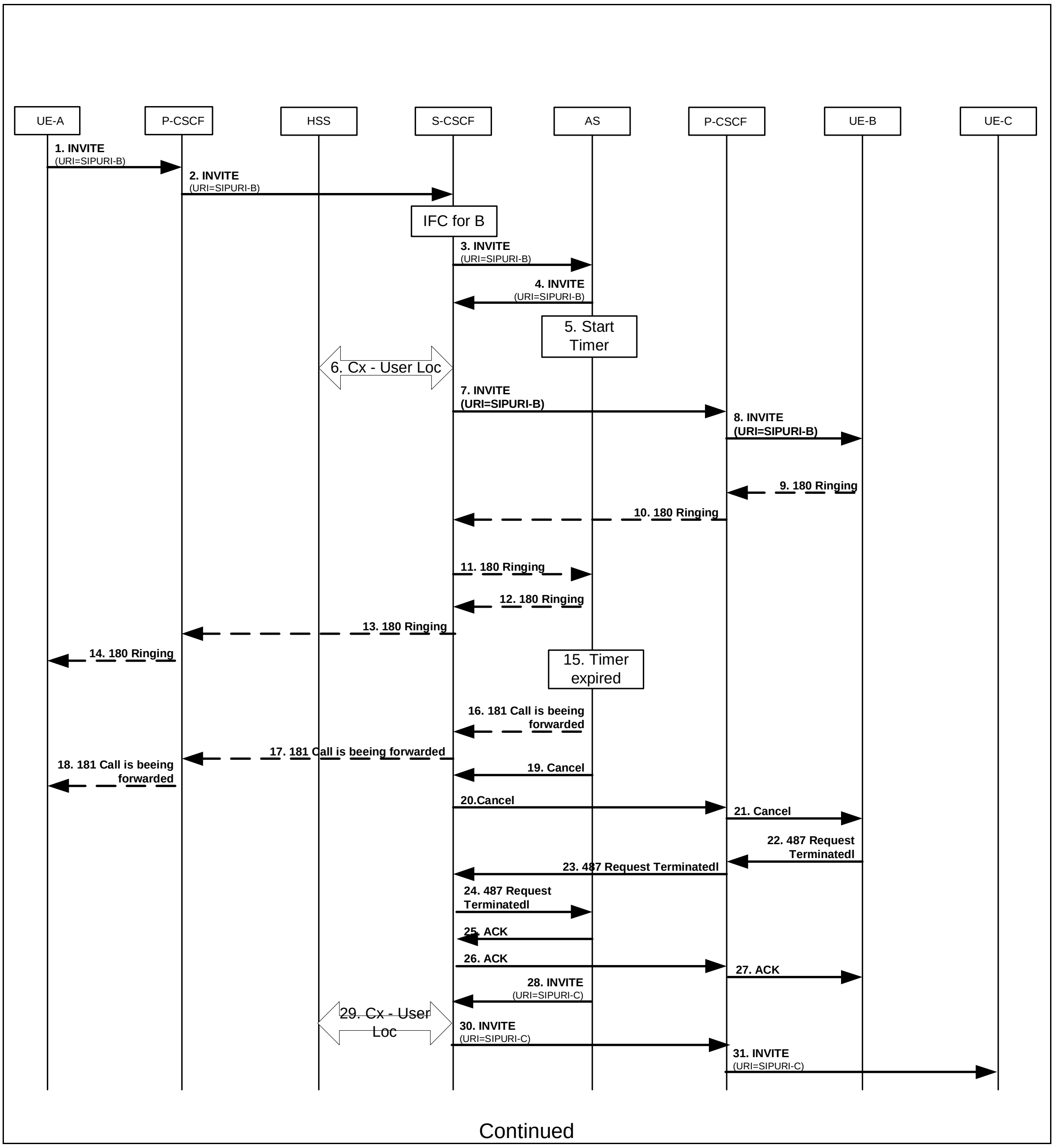
SSS__XXSSCFNR 05	**NGN reference to: [17]**
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNR
Configuration	The user A and the user C and D are in network N1. The user B is in network N2 and is provided with CFNR. The user A and the user C are in network N1 and user C is provided with TIP. The user 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; Case c) Supported: 100 rel and precondition.
Comments:	
Continued:	

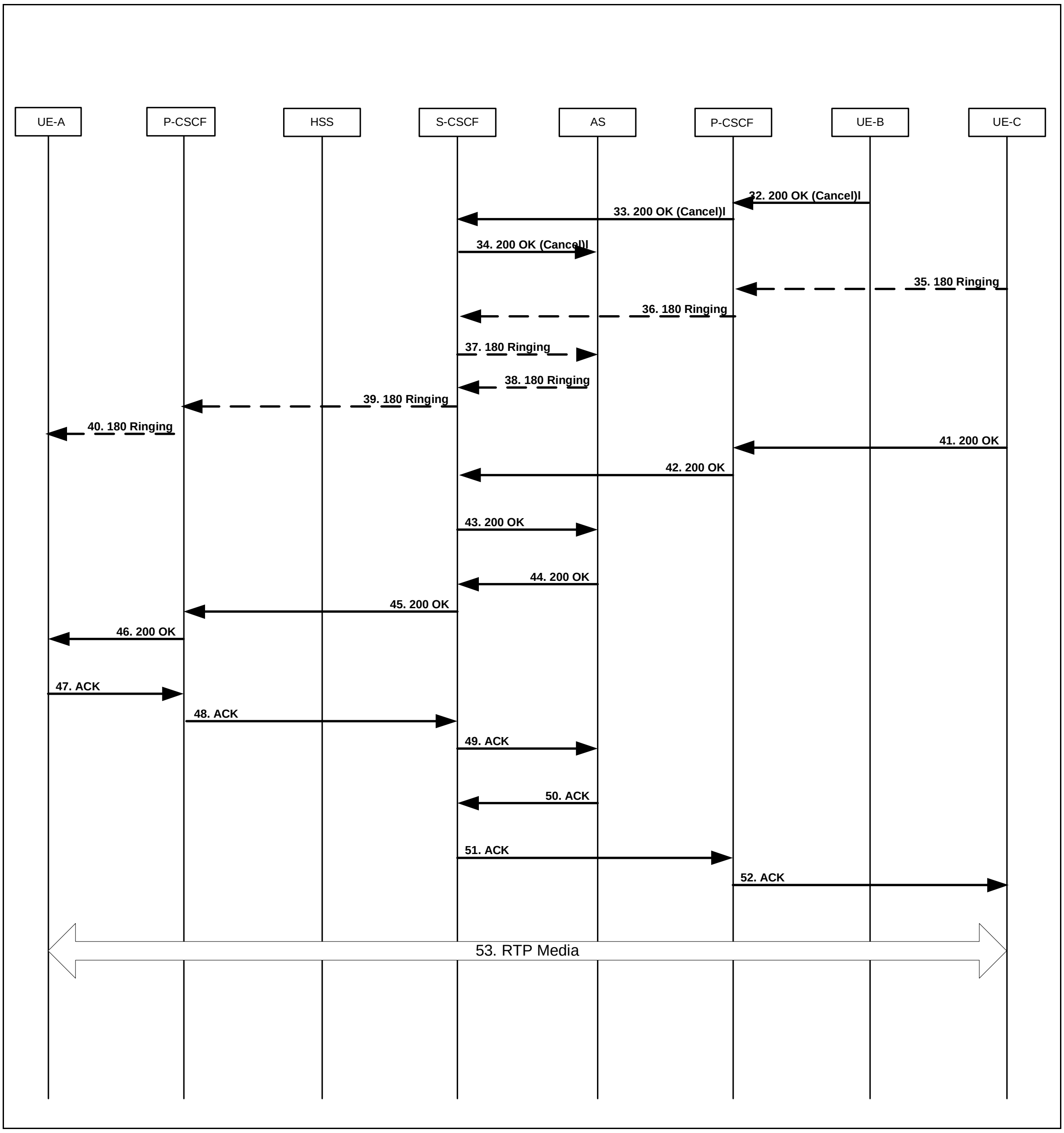

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
1. INVITE (URI=SIPURI-B)
2. INVITE (URI=SIPURI-B)
IFC for B
3. INVITE (URI=SIPURI-B)
4. INVITE (URI=SIPURI-B)
5. Start Timer
6. Cx - User Loc
7. INVITE (URI=SIPURI-B)
8. INVITE (URI=SIPURI-B)
9. 180 Ringing
10. 180 Ringing
11. 180 Ringing
12. 180 Ringing
13. 180 Ringing
14. 180 Ringing
15. Timer expired
16. 181 Call is beeing forwarded
17. 181 Call is beeing forwarded
18. 181 Call is beeing forwarded
19. Cancel
20.Cancel
21. Cancel
22. 487 Request Terminatedl
23. 487 Request Terminatedl
24. 487 Request Terminatedl
25. ACK
26. ACK
27. ACK
28. INVITE (URI=SIPURI-C)
29. Cx - User Loc
30. INVITE (URI=SIPURI-C)
31. INVITE (URI=SIPURI-C)
Continued


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
32. 200 OK (Cancel)l
33. 200 OK (Cancel)l
34. 200 OK (Cancel)l
35. 180 Ringing
36. 180 Ringing
37. 180 Ringing
38. 180 Ringing
39. 180 Ringing
40. 180 Ringing
41. 200 OK
42. 200 OK
43. 200 OK
44. 200 OK
45. 200 OK
46. 200 OK
47. ACK
48. ACK
49. ACK
50. ACK
51. ACK
52. ACK
53. RTP Media

SSS__XXSSCFNR 06	**NGN reference to: [17]**	
TSS reference:	SIP-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: o 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:		

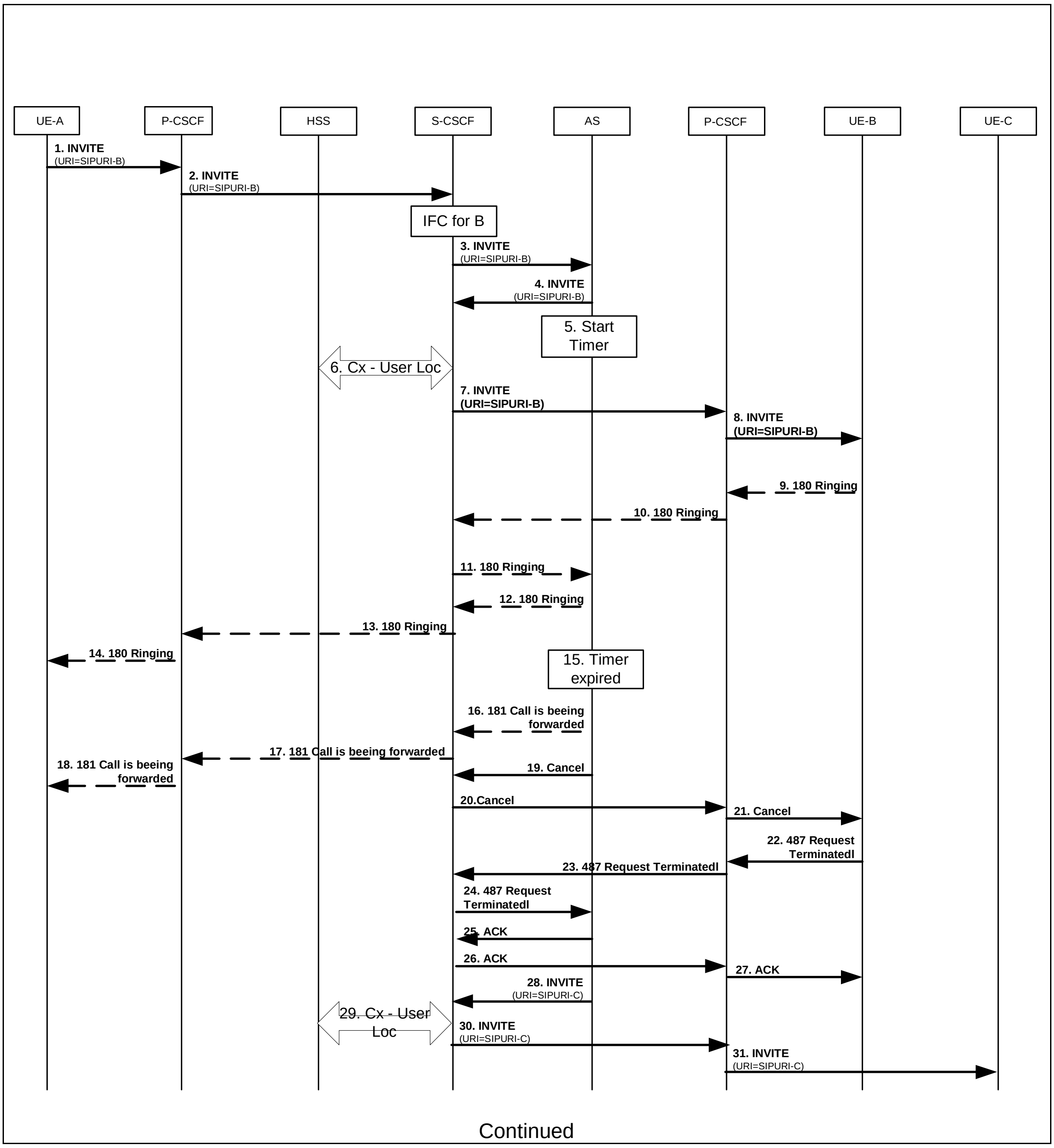
Continued

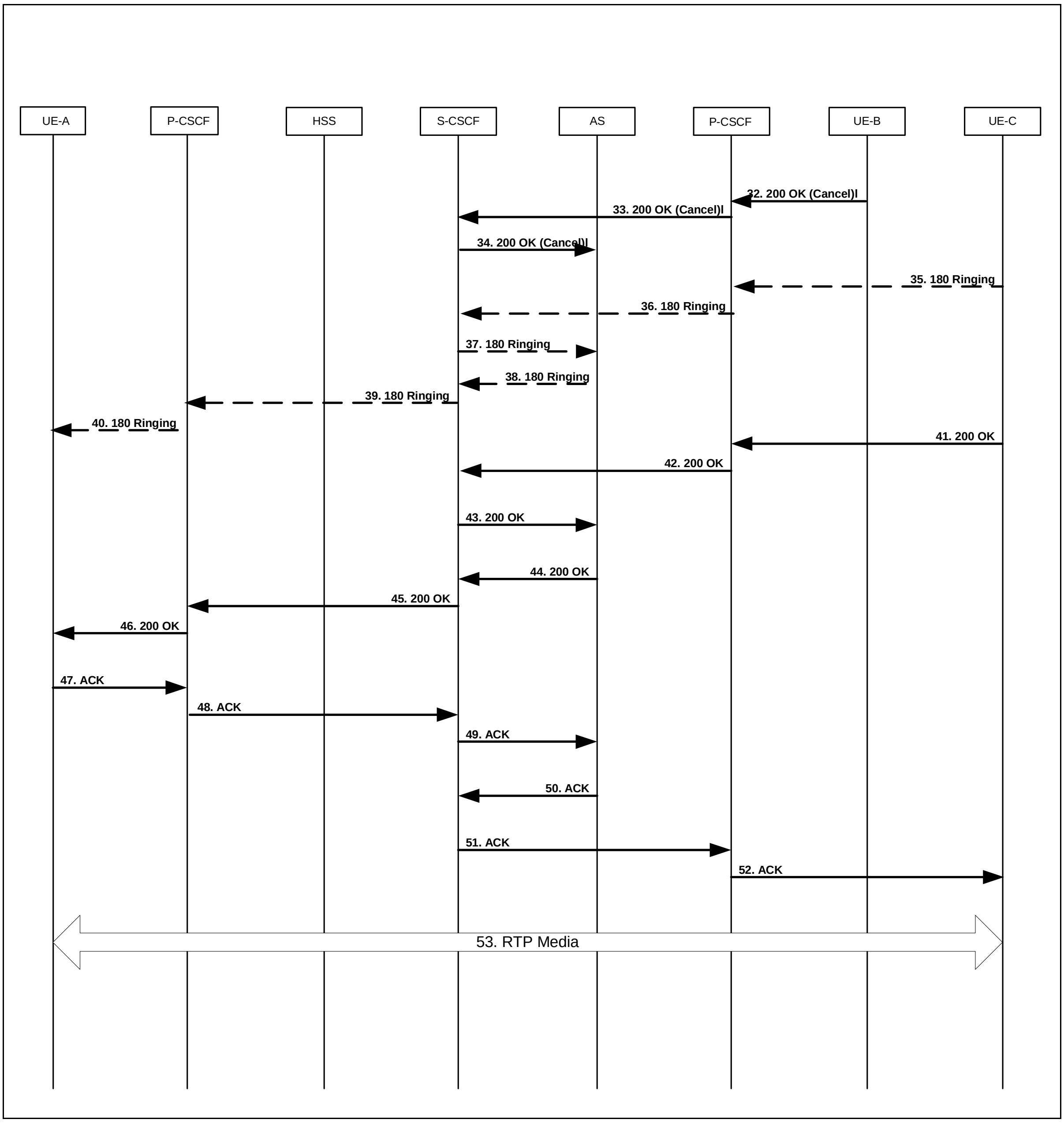

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
1. INVITE (URI=SIPURI-B)
2. INVITE (URI=SIPURI-B)
IFC for B
3. INVITE (URI=SIPURI-B)
4. INVITE (URI=SIPURI-B)
5. Start Timer
6. Cx - User Loc
7. INVITE (URI=SIPURI-B)
8. INVITE (URI=SIPURI-B)
9. 180 Ringing
10. 180 Ringing
11. 180 Ringing
12. 180 Ringing
13. 180 Ringing
14. 180 Ringing
15. Timer expired
16. 181 Call is beeing forwarded
17. 181 Call is beeing forwarded
18. 181 Call is beeing forwarded
19. Cancel
20.Cancel
21. Cancel
22. 487 Request Terminatedl
23. 487 Request Terminatedl
24. 487 Request Terminatedl
25. ACK
26. ACK
27. ACK
28. INVITE (URI=SIPURI-C)
29. Cx - User Loc
30. INVITE (URI=SIPURI-C)
31. INVITE (URI=SIPURI-C)
Continued


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
32. 200 OK (Cancel)l
33. 200 OK (Cancel)l
34. 200 OK (Cancel)l
35. 180 Ringing
36. 180 Ringing
37. 180 Ringing
38. 180 Ringing
39. 180 Ringing
40. 180 Ringing
41. 200 OK
42. 200 OK
43. 200 OK
44. 200 OK
45. 200 OK
46. 200 OK
47. ACK
48. ACK
49. ACK
50. ACK
51. ACK
52. ACK
53. RTP Media

SSS__XXSSCFNR 07	NGN reference to: [17]	
TSS reference:	SIP-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: o 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 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:		

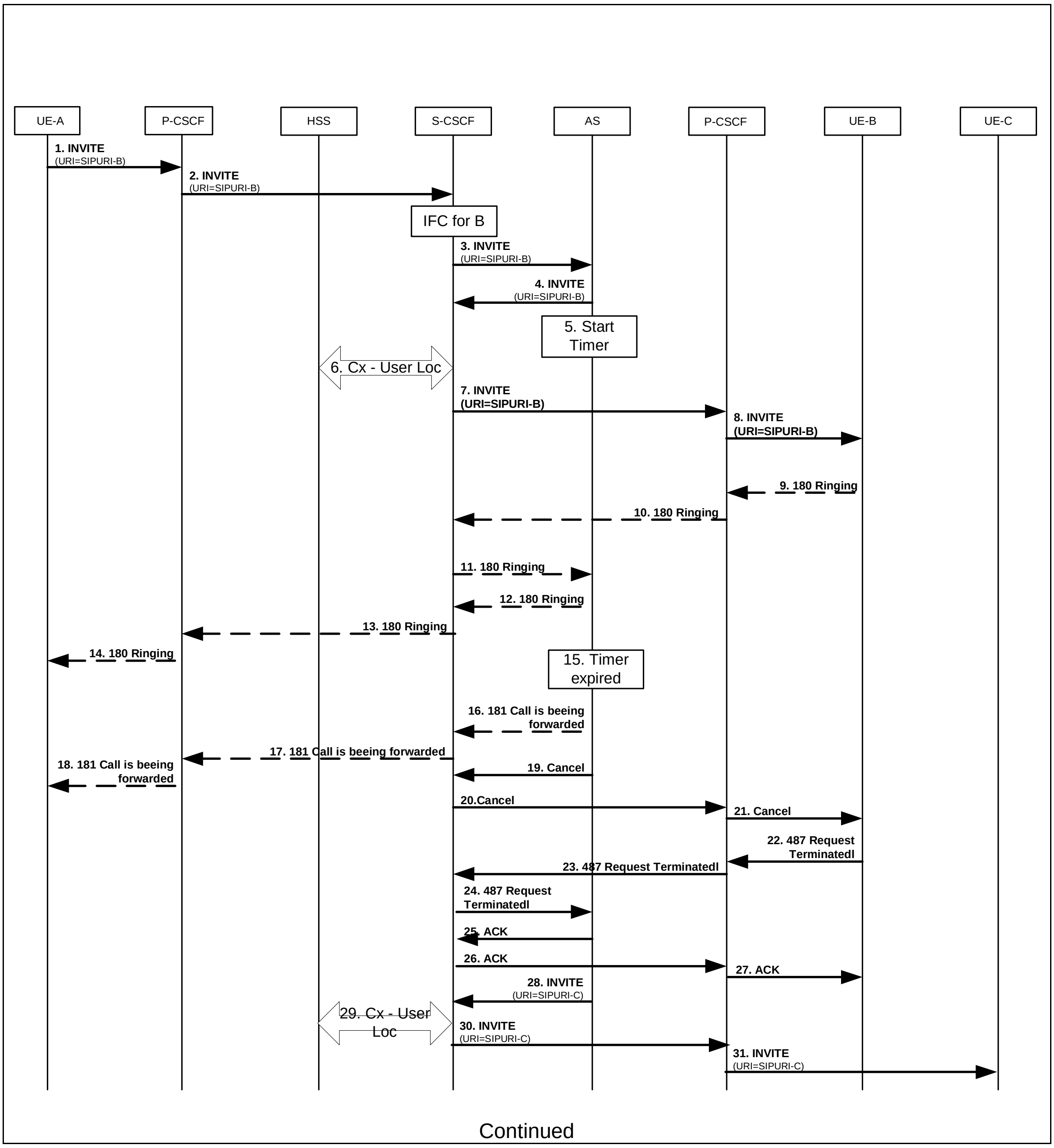
Continued

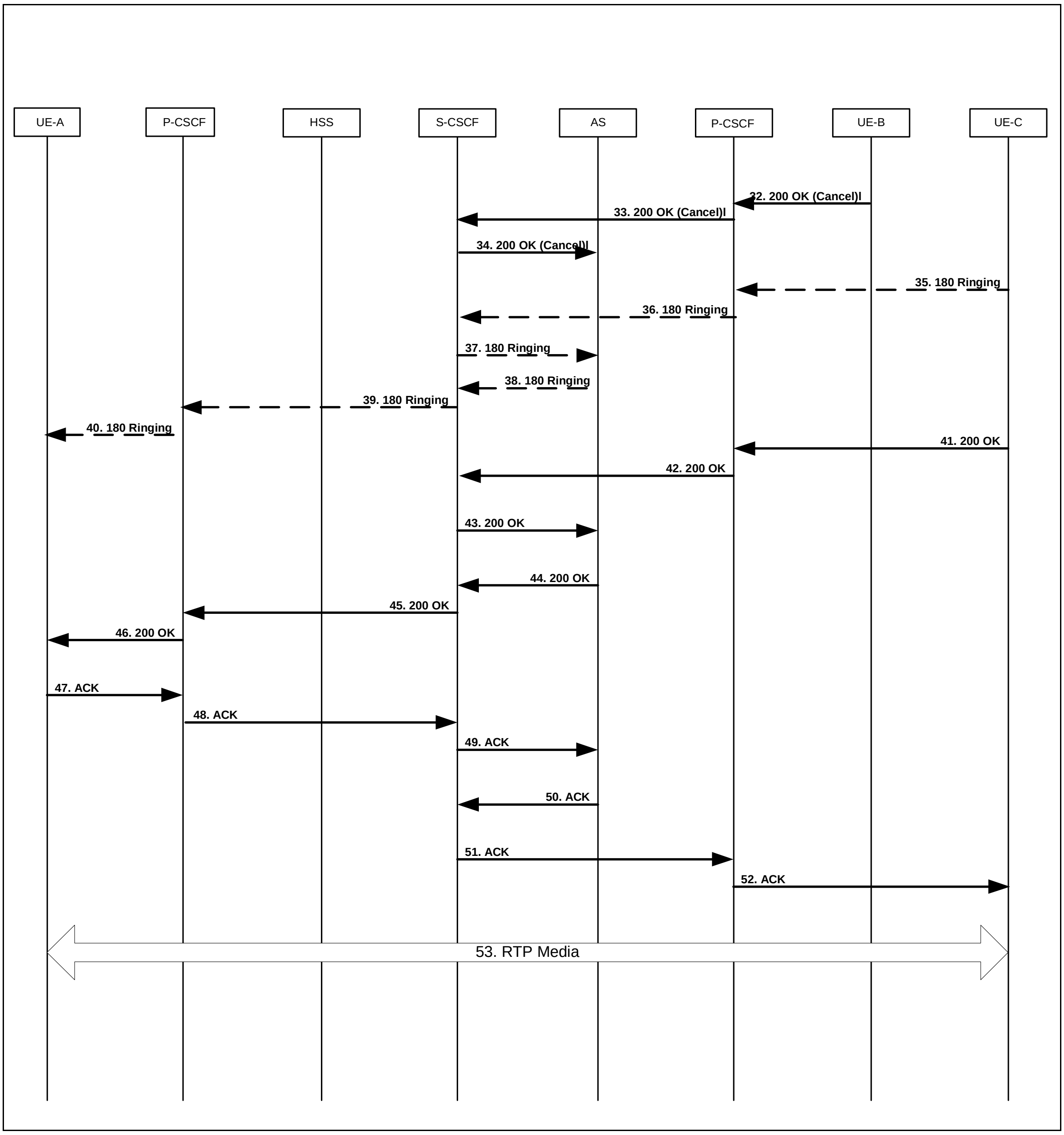

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
1. INVITE
(URI=SIPURI-B)
2. INVITE
(URI=SIPURI-B)
IFC for B
3. INVITE
(URI=SIPURI-B)
4. INVITE
(URI=SIPURI-B)
5. Start Timer
6. Cx - User Loc
7. INVITE
(URI=SIPURI-B)
8. INVITE
(URI=SIPURI-B)
9. 180 Ringing
10. 180 Ringing
11. 180 Ringing
12. 180 Ringing
13. 180 Ringing
14. 180 Ringing
15. Timer expired
16. 181 Call is beeing forwarded
17. 181 Call is beeing forwarded
18. 181 Call is beeing forwarded
19. Cancel
20.Cancel
21. Cancel
22. 487 Request Terminatedl
23. 487 Request Terminatedl
24. 487 Request Terminatedl
25. ACK
26. ACK
27. ACK
28. INVITE
(URI=SIPURI-C)
29. Cx - User Loc
30. INVITE
(URI=SIPURI-C)
31. INVITE
(URI=SIPURI-C)
Continued


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
32. 200 OK (Cancel)I
33. 200 OK (Cancel)I
34. 200 OK (Cancel)I
35. 180 Ringing
36. 180 Ringing
37. 180 Ringing
38. 180 Ringing
39. 180 Ringing
40. 180 Ringing
41. 200 OK
42. 200 OK
43. 200 OK
44. 200 OK
45. 200 OK
46. 200 OK
47. ACK
48. ACK
49. ACK
50. ACK
51. ACK
52. ACK
53. RTP Media

SSS__XXSSCFNR 08	**NGN reference to: [17]**	
TSS reference:	SIP-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: o 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

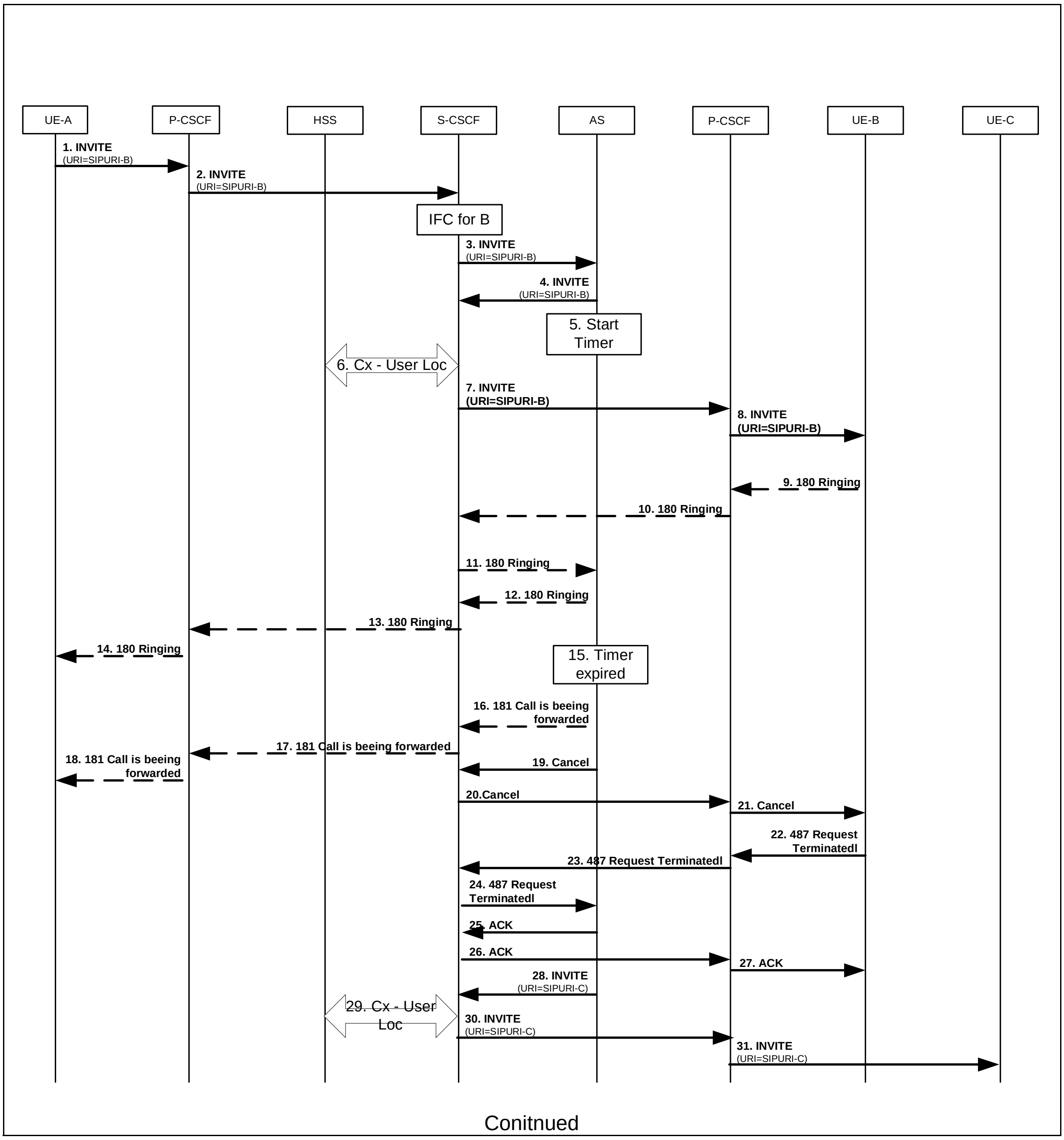

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
1. INVITE
(URI=SIPURI-B)
2. INVITE
(URI=SIPURI-B)
IFC for B
3. INVITE
(URI=SIPURI-B)
4. INVITE
(URI=SIPURI-B)
5. Start Timer
6. Cx - User Loc
7. INVITE
(URI=SIPURI-B)
8. INVITE
(URI=SIPURI-B)
9. 180 Ringing
10. 180 Ringing
11. 180 Ringing
12. 180 Ringing
13. 180 Ringing
14. 180 Ringing
15. Timer expired
16. 181 Call is beeing forwarded
17. 181 Call is beeing forwarded
18. 181 Call is beeing forwarded
19. Cancel
20.Cancel
21. Cancel
22. 487 Request Terminatedl
23. 487 Request Terminatedl
24. 487 Request Terminatedl
25. ACK
26. ACK
27. ACK
28. INVITE
(URI=SIPURI-C)
29. Cx - User Loc
30. INVITE
(URI=SIPURI-C)
31. INVITE
(URI=SIPURI-C)
Continued

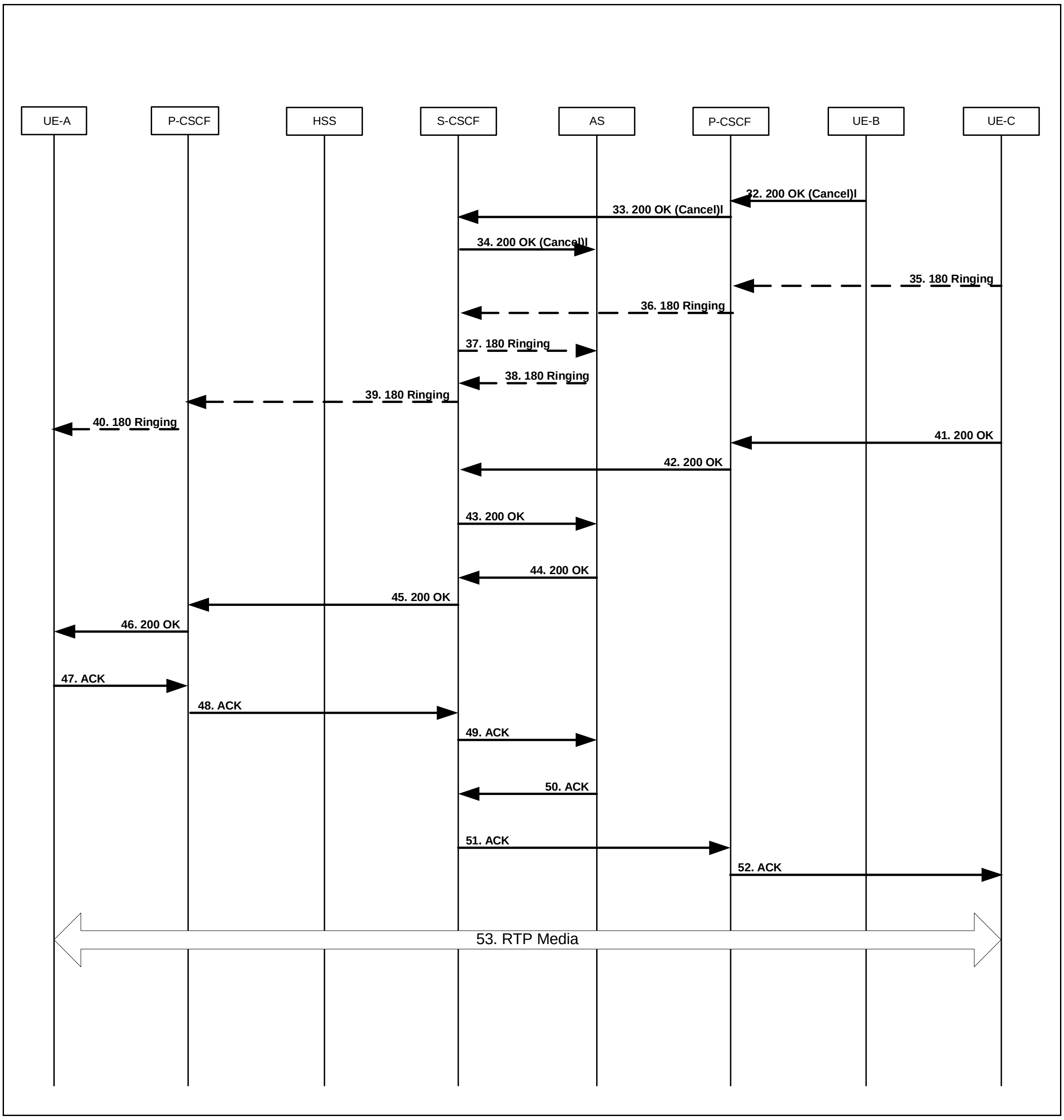

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
32. 200 OK (Cancel)I
33. 200 OK (Cancel)I
34. 200 OK (Cancel)I
35. 180 Ringing
36. 180 Ringing
37. 180 Ringing
38. 180 Ringing
39. 180 Ringing
40. 180 Ringing
41. 200 OK
42. 200 OK
43. 200 OK
44. 200 OK
45. 200 OK
46. 200 OK
47. ACK
48. ACK
49. ACK
50. ACK
51. ACK
52. ACK
53. RTP Media

SSS__XXSSCFNR 09	**NGN reference to: [17]**
TSS reference:	SIP-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-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 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


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
1. INVITE (URI=SIPURI-B)
2. INVITE (URI=SIPURI-B)
IFC for B
3. INVITE (URI=SIPURI-B)
4. INVITE (URI=SIPURI-B)
5. Start Timer
6. Cx - User Loc
7. INVITE (URI=SIPURI-B)
8. INVITE (URI=SIPURI-B)
9. 180 Ringing
10. 180 Ringing
11. 180 Ringing
12. 180 Ringing
13. 180 Ringing
14. 180 Ringing
15. Timer expired
16. 181 Call is beeing forwarded
17. 181 Call is beeing forwarded
18. 181 Call is beeing forwarded
19. Cancel
20.Cancel
21. Cancel
22. 487 Request Terminatedl
23. 487 Request Terminatedl
24. 487 Request Terminatedl
25. ACK
26. ACK
27. ACK
28. INVITE (URI=SIPURI-C)
29. Cx - User Loc
30. INVITE (URI=SIPURI-C)
31. INVITE (URI=SIPURI-C)
Conitnued



SSS__XXSSCFNR 15	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 [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 busy.	
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=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 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 network. • CFNR supported. • user C is network determined user busy.	
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=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 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-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, 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:	The served user subscribes to the CFNR simulation service; originating user is notified. Additionaly an announcement is played. When Communication Diversion occurs and if the notification procedures of the originating user is supported then a 181 (Call Is Being Forwarded) response shall be sent towards the originating user. Additional the AS initiates an announcement to be included towards the calling user in order to inform about the diversion. Announcements shall be played according to procedures as are described in TS 183 028 [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 **SIP header values:** 181 Call is Being Forwarded:
Comments:	

SIP#1		**SUT**			**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE					
			INVITE	→	INVITE		
180 Ringing	←		180 Ringing	←	180 Ringing		
			CANCEL	→	CANCEL		
181 Call is Being Forwarded	←	181 Call is Being Forwarded					
			INVITE	→		→	INVITE
			180 Ringing	←		←	180 Ringing
180 Ringing	←	180 Ringing					
			200 OK (INVITE)	←		←	200 OK (INVITE)
200 OK (INVITE)	←	200 OK (INVITE)					
ACK	→					→	ACK
		Communication					
BYE	→					→	BYE
200 OK (BYE)	←					←	200 OK (BYE)

SSS__XXSSCFNR 19	**NGN reference to: [17] clause 4.5.2.6.3**			
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 network determined user busy.			
Test purpose:	The served user subscribes to the CFNR simulation service; Indication of communication diversion to the diverting user using the MESSAGE request triggered by a timer. Ensure that when the diverting user is registering the AS sends a MESSAGE request including the information where the call is diverted to. The MESSAGE request that is be sent due to an timer value that can be provided by the 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:** 181 Call is Being Forwarded:			
Comments:				
SIP#1		**SUT**	**SIP#2 (served user)**	**SIP#3**
INVITE 1	→	INVITE		
		INVITE →	INVITE	
180 Ringing	←	180 Ringing ←	180 Ringing	
		CANCEL →	CANCEL	
		INVITE →		→ INVITE
		MESSAGE →	MESSAGE	
		200 OK ←	200 OK	
		180 Ringing ←		← 180 Ringing
180 Ringing	←	180 Ringing		
		200 OK (INVITE) ←		← 200 OK (INVITE)
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→			→ ACK
		Communication		
BYE	→			→ BYE
200 OK (BYE)	←			← 200 OK (BYE)

SSS__XXSSCFNR 21	**NGN reference to: [17] clause 4.5.2.6.3**
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 network determined user busy.
Test purpose:	*The served user subscribes to the CFNR simulation service; Indication of communication diversion to the diverting user using the MESSAGE request when a new outgoing communication is requested.* Ensure that a diverting user will be informed with a MESSAGE request after the diverting user has initiated a new outgoing communication the information where the 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; Case c) Supported: 100 rel and precondition. **SIP header values:** 181 Call is Being Forwarded:

Comments:

SIP#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE				
		INVITE	→	INVITE		
180 Ringing	←	180 Ringing	←	180 Ringing		
		CANCEL	→	CANCEL		
		INVITE	→		→	INVITE
		MESSAGE	→	MESSAGE		
		200 OK	→	200 OK		
		180 Ringing	←		←	180 Ringing
180 Ringing	←	180 Ringing				
		200 OK (INVITE)	←		←	200 OK (INVITE)
200 OK (INVITE)	←	200 OK (INVITE)				
ACK	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)
Communication						
		INVITE 3	→		→	INVITE
		180 Ringing	←		←	180 Ringing
			←	180 Ringing		
			←	200 OK (INVITE)	←	200 OK (INVITE)
		MESSAGE	→	MESSAGE		
		200 OK	→	200 OK		
			→	ACK	→	ACK

SSS__XXSSCFNR 22	**NGN reference to: [17] clause 4.5.2.6.3**	
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 network determined user busy.	
Test purpose:	The served user subscribes to the CFNR simulation service; Indication of communication diversion to the diverting user using a voicemail or mail system. Ensure that when communication diversion is performed, the AS sends Voicemail or Message mail system including the information where the 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; Case c) Supported: 100 rel and precondition.	

Comments:

SIP#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE				
		INVITE	→	INVITE		
180 Ringing	←	180 Ringing	←	180 Ringing		
		CANCEL	→	CANCEL		
		INVITE	→		→	INVITE
		Voicemail or Message mail system	→	**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	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)

The test case can not be tested with an ATS

SSS__XXSSCFNR 23	**NGN reference to: [17] clause 4.5.2.6.3**
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 network determined user busy.
Test purpose:	The served user subscribes to the CFNR simulation service; Indication of communication diversion to the diverting user using a voicemail or mail system requested by a timer. Ensure that when communication diversion is performed, the AS sends a Voicemail or Message mail system request including the information where the call is diverted to. The Voicemail or Message mail system shall be sent due to a timer value that can be provided by the 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#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE				
		INVITE	→	INVITE		
180 Ringing	←	180 Ringing	←	180 Ringing		
		CANCEL	→	CANCEL		
		INVITE	→		→	INVITE
		180 Ringing	←		←	180 Ringing
180 Ringing	←	180 Ringing				
		200 OK (INVITE)	←		←	200 OK (INVITE)
200 OK (INVITE)	←	200 OK (INVITE)				
		Voicemail or Message mail system	→	**Voicemail or Message mail system**		
ACK	→				→	ACK
		Communication				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)

The test case can not be tested with an ATS

SSS__XXSSCFNR 24	**NGN reference to: [17] clause 4.5.2.6.3**	
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 network determined user busy.	
Test purpose:	The served user subscribes to the CFNR simulation service; periodically indication of communication diversion to the diverting user using a voicemail or mail system. Ensure that when communication diversion is performed. the AS sends a Voicemail or Message mail system request including the information where the call is diverted to. The diverting user will be informed periodically with Voicemail or Message mail system with the information where the 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 Case c) Supported: 100 rel and precondition	

Comments:

SIP#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE				
		INVITE	→	INVITE		
180 Ringing	←	180 Ringing	←	180 Ringing		
		CANCEL	→	CANCEL		
		INVITE	→		→	INVITE
		Voicemail or Message mail system	→	**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	→				→	ACK
		Voicemail or Message mail system		**Voicemail or Message mail system**		
		Communication				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)

The test case can not be tested with an ATS

SSS__XXSSCFNR 25	**NGN reference to: [17] clause 4.5.2.6.3**
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 network determined user busy.
Test purpose:	The served user subscribes to the CFNR simulation service; Indication of a new communication diversion to the diverting user using a voicemail or mail system. Ensure that a diverting user will be informed with Message Voicemail or Message mail system after the diverting user has initiated a new outgoing communication the information where the 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 Case c) Supported: 100 rel and precondition

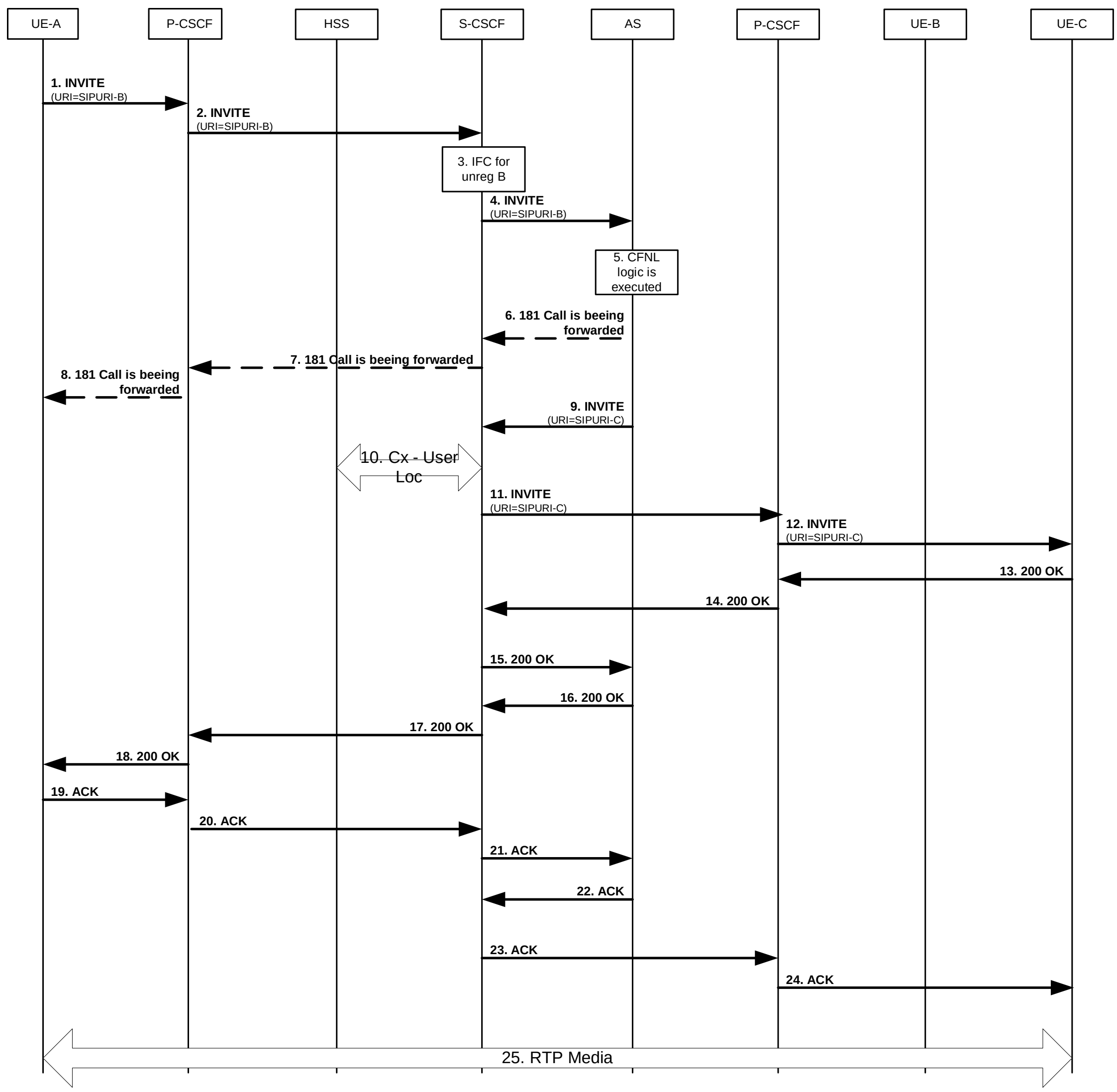
Comments:

SIP#1		**SUT**		**SIP#2 (served user)**		**SIP#3**
INVITE 1	→	INVITE				
		INVITE	→	INVITE		
180 Ringing	←	180 Ringing	←	180 Ringing		
		CANCEL	→	CANCEL		
		INVITE	→		→	INVITE
		Voicemail or Message mail system	→	**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	→				→	ACK
		Communication				
		INVITE	→		→	INVITE
		Voicemail or Message mail system	→	**Voicemail or Message mail system**		
				180 Ringing	←	180 Ringing
				200 OK (INVITE)	←	200 OK (INVITE)
				ACK	→	ACK

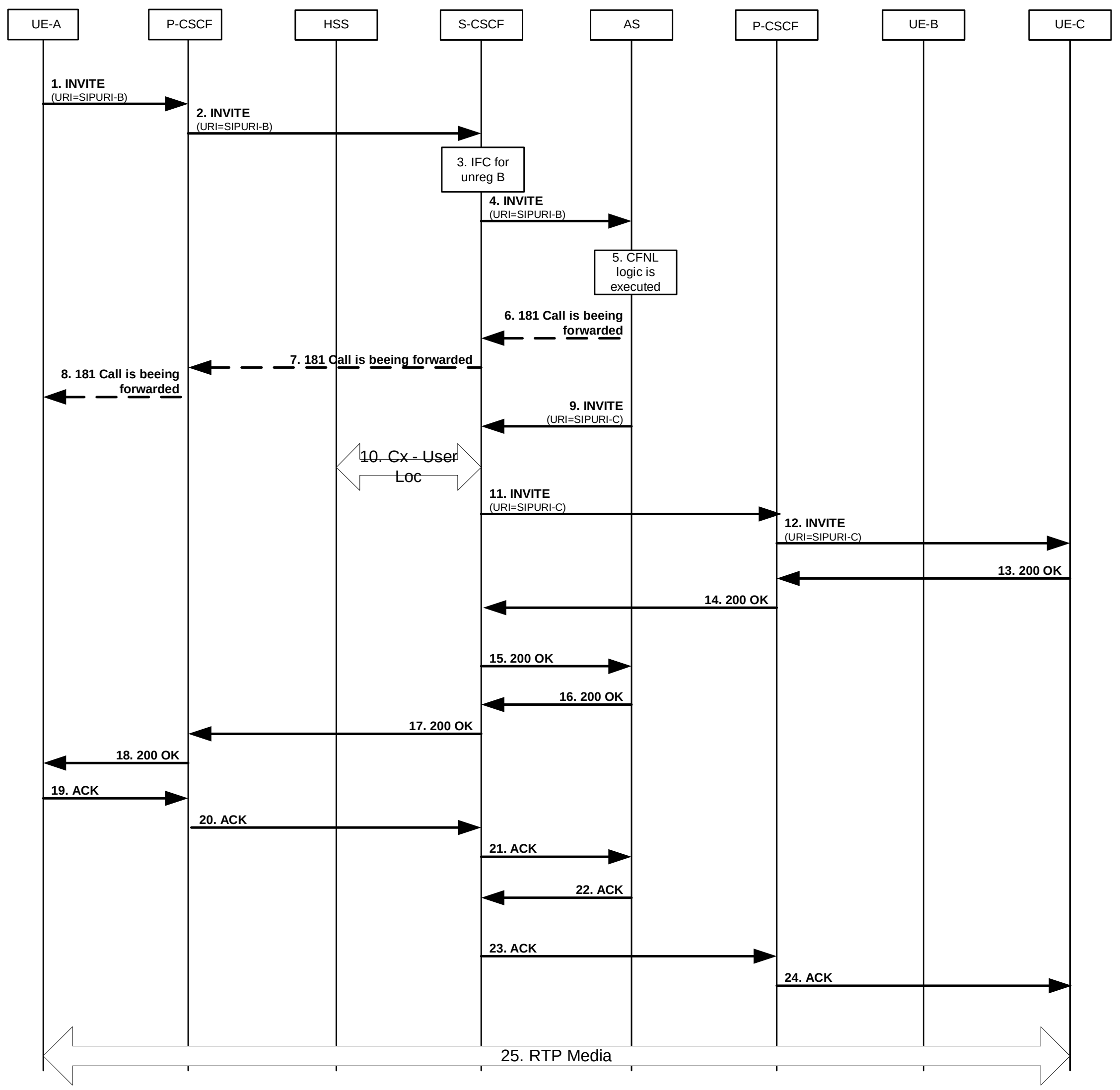
The test case can not be tested with an ATS

6.5.2.9 CFNL

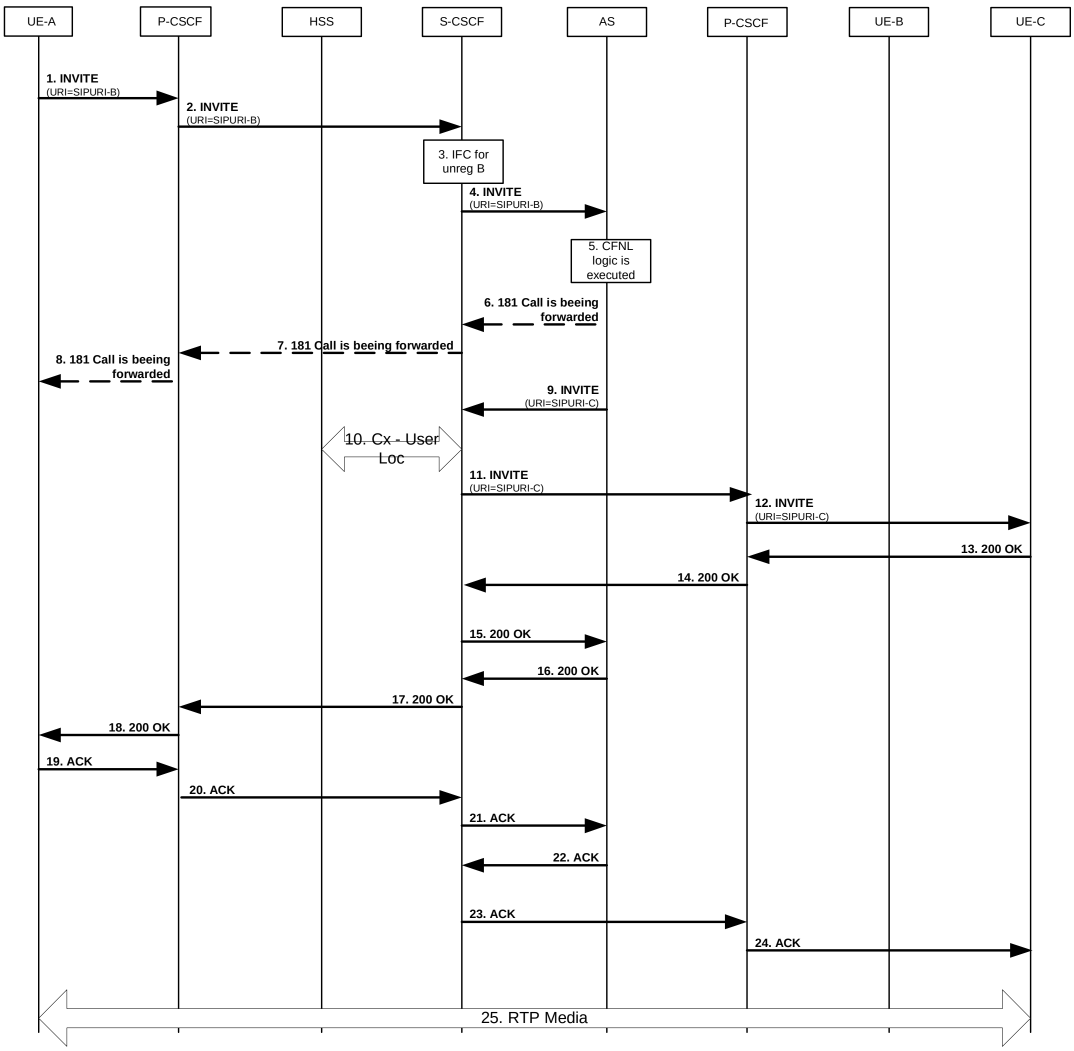
SSS__XXSSCFNL 01	NGN reference to: [17]	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CFNL	
Configuration	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNL	
Selection criteria:	• Call forwarding by the network: o CFNL supported.	
Test purpose:	Ensure that when user A calls user B, the call is forwarded to user C. Ensure that in the active call state the voice transfer on the media channels is performed correctly (e.g. testing QoS parameters).	
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:		



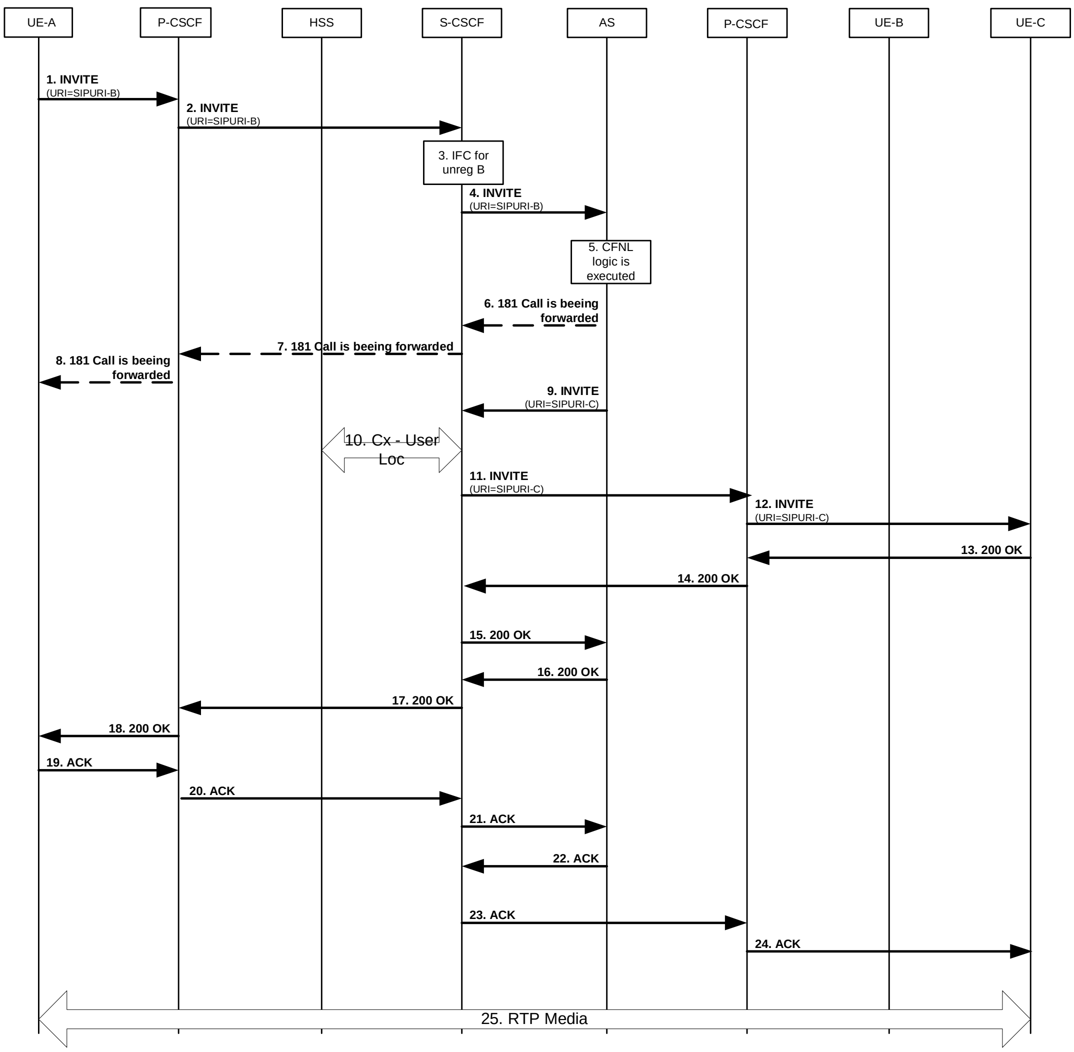
SSS__XXSSCFNL 02	**NGN reference to: [17]**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNL.	
Configuration	The user A and the user C are in network N1. The user B is provided with CFNL. ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting number is released to the diverted-to user"=Yes).	
Selection criteria:	• Call forwarding by the network. • CFNL 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 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).	
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:		



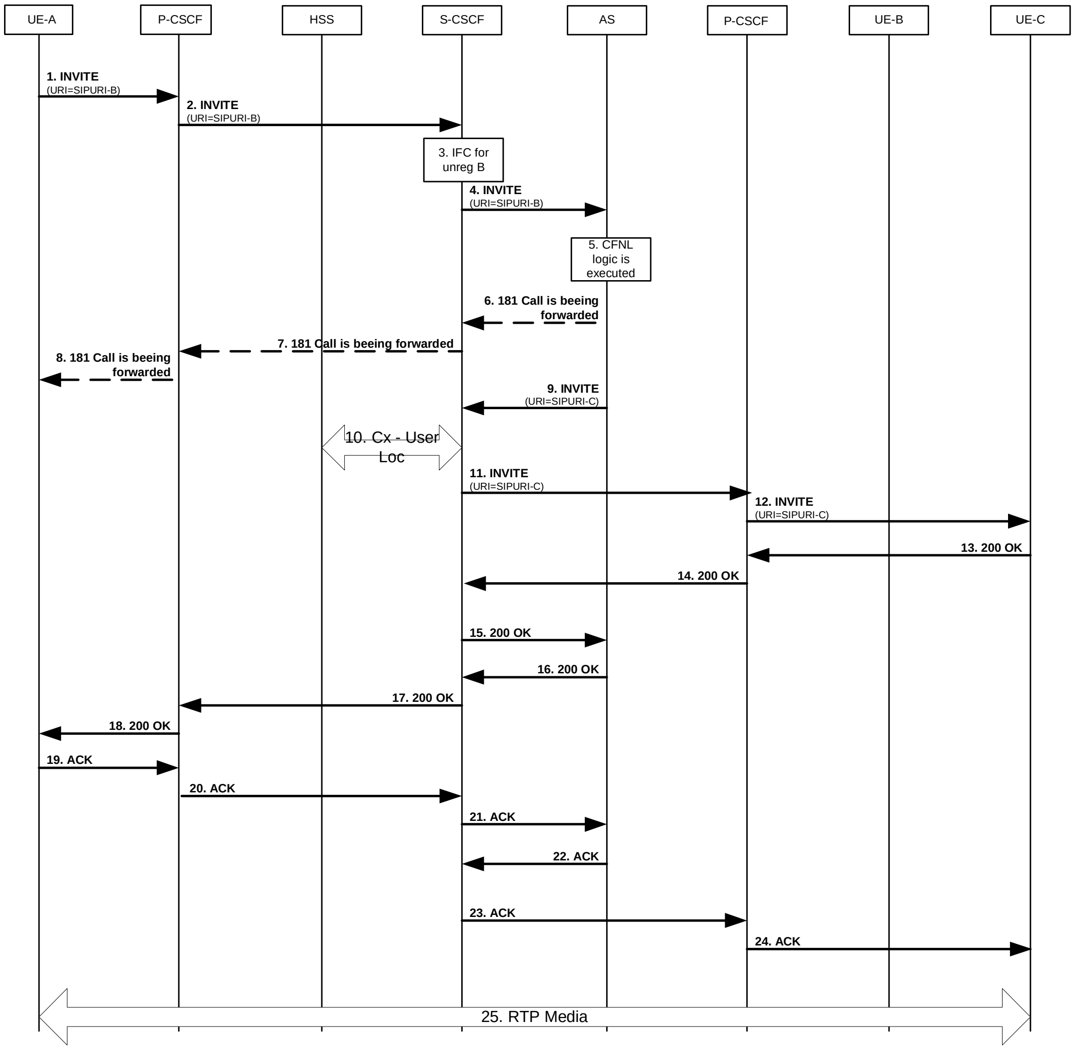
SSS__XXSSCFNL 03	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNL
Configuration	The user A and the user C are in network N1. The user B is in network N2 and is provided with CFNL ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting number is released to the diverted-to user"=No).
Selection criteria:	• Call forwarding by the network. • CFNL 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:	



SSS__XXSSCFNL 04	**NGN reference to: [17]**
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNL.
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 CFNL. 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 CFNL ("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).
Selection criteria:	• Call forwarding by the network. • CFNL 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.
Comments:	



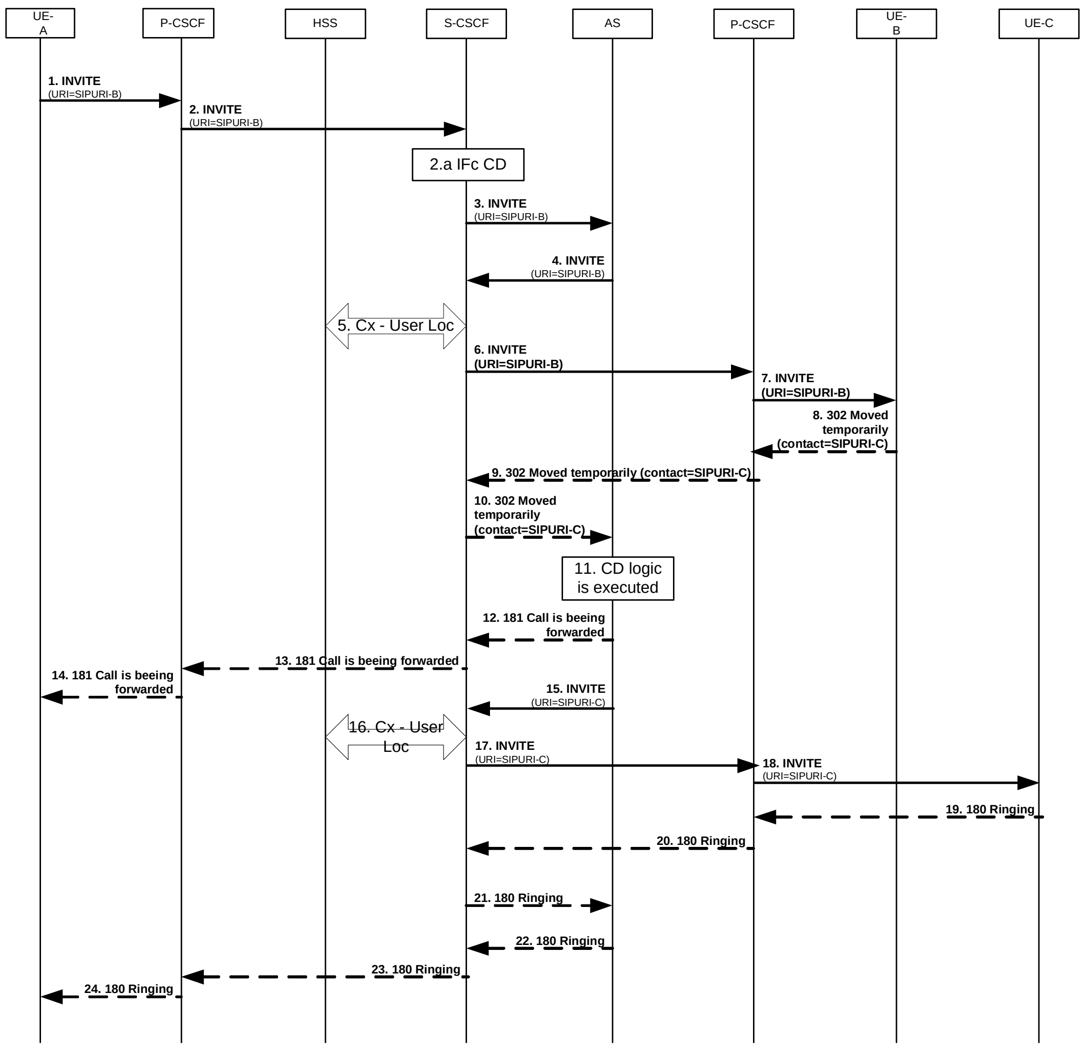
SSS__XXSSCFNL 05	IMS reference to: [17]	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNL.	
Configuration	The user B is in network N2 and is provided with CFNL.	
Selection criteria:	• Call forwarding by the network. • CFNL supported. • user C is user determined user busy.	
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 **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:		



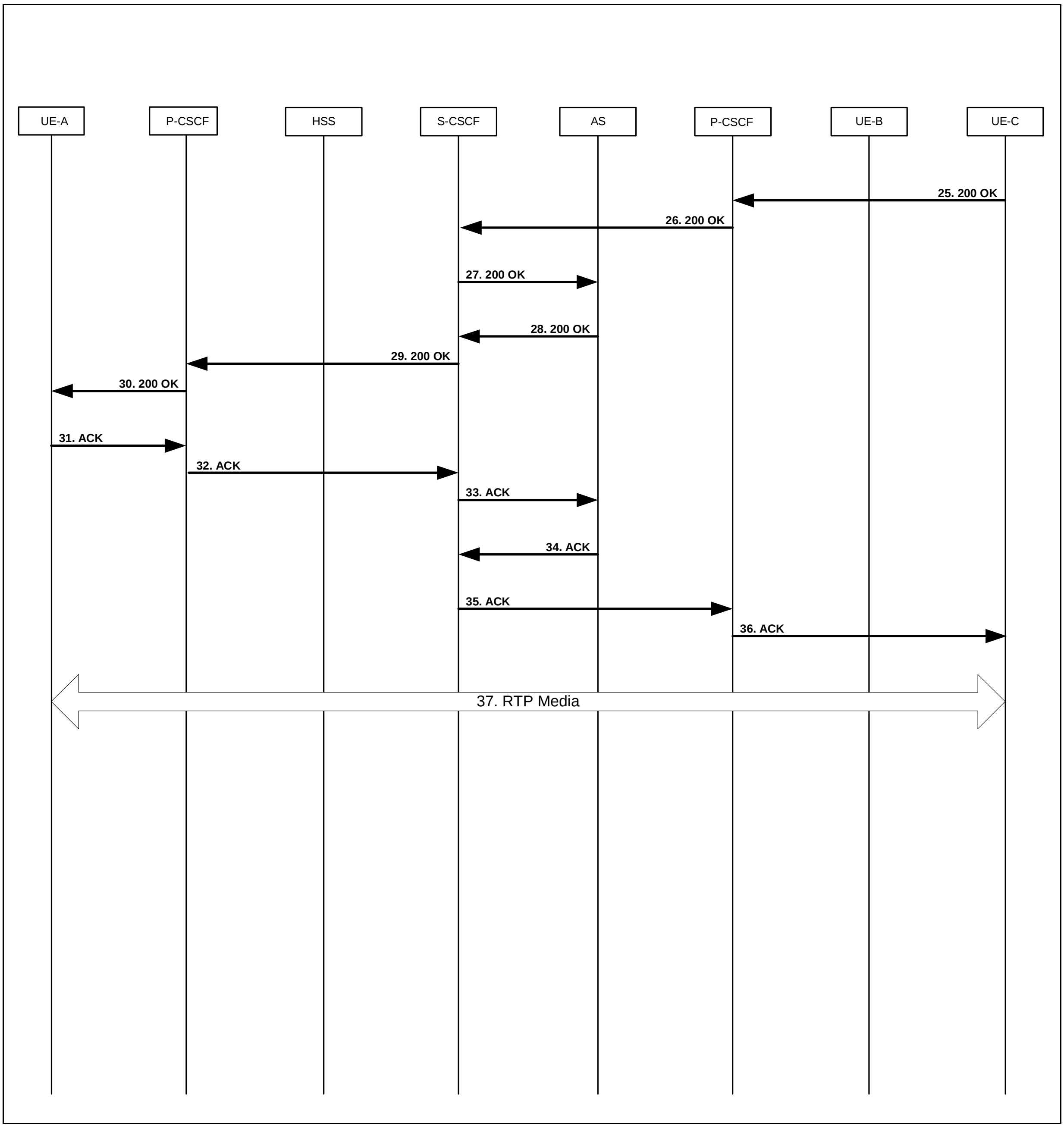
SSS__XXSSCFNL 06	**NGN reference to: [17]**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/CFNL.	
Configuration	The user B is in network N2 and is provided with CFNL.	
Selection criteria:	• Call forwarding by the network. • CFNL supported. • user C is network determined user busy.	
Test purpose:	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	The user A and the user C are in network N1. The user B is in network N2 and is provided with CD- Immediate response.	
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.	
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:		

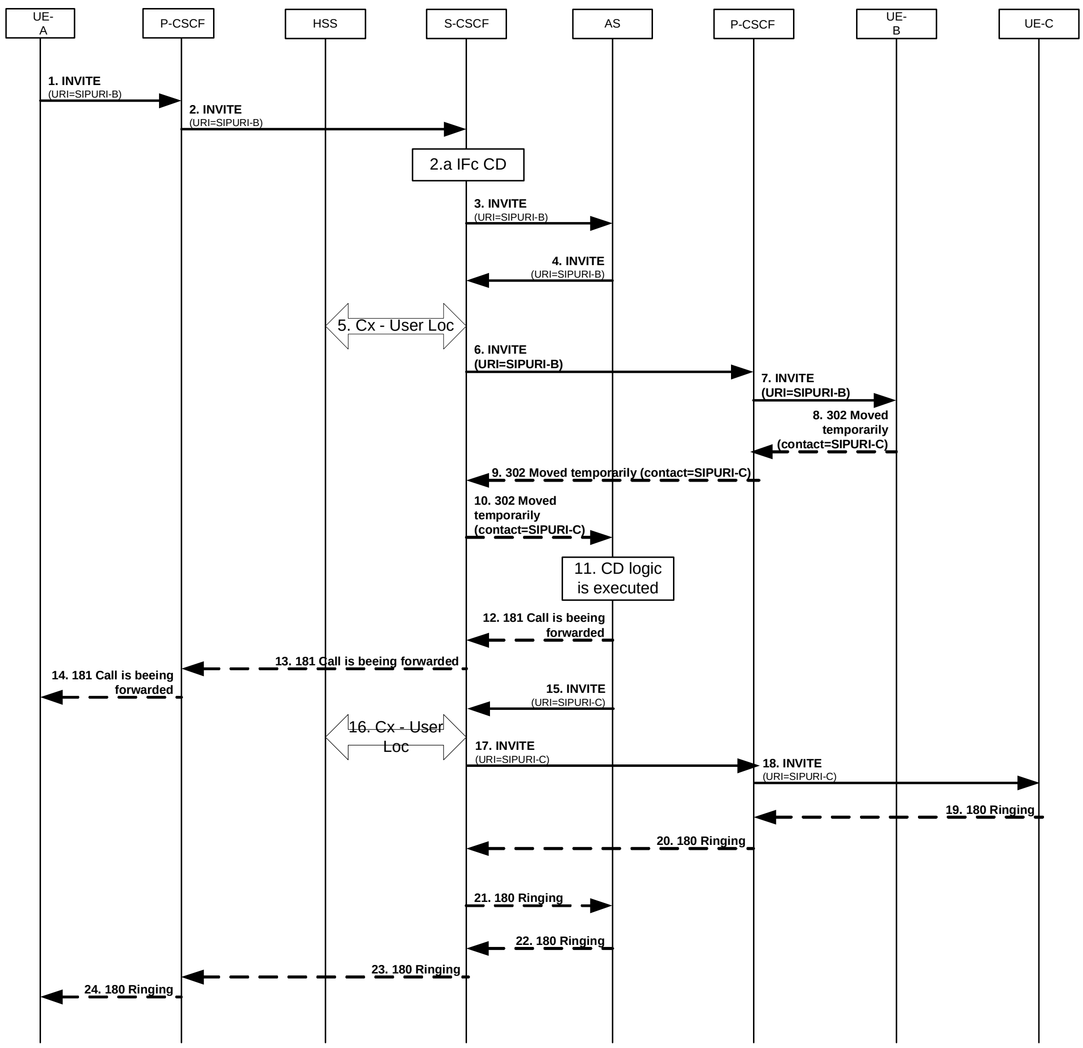


Continued

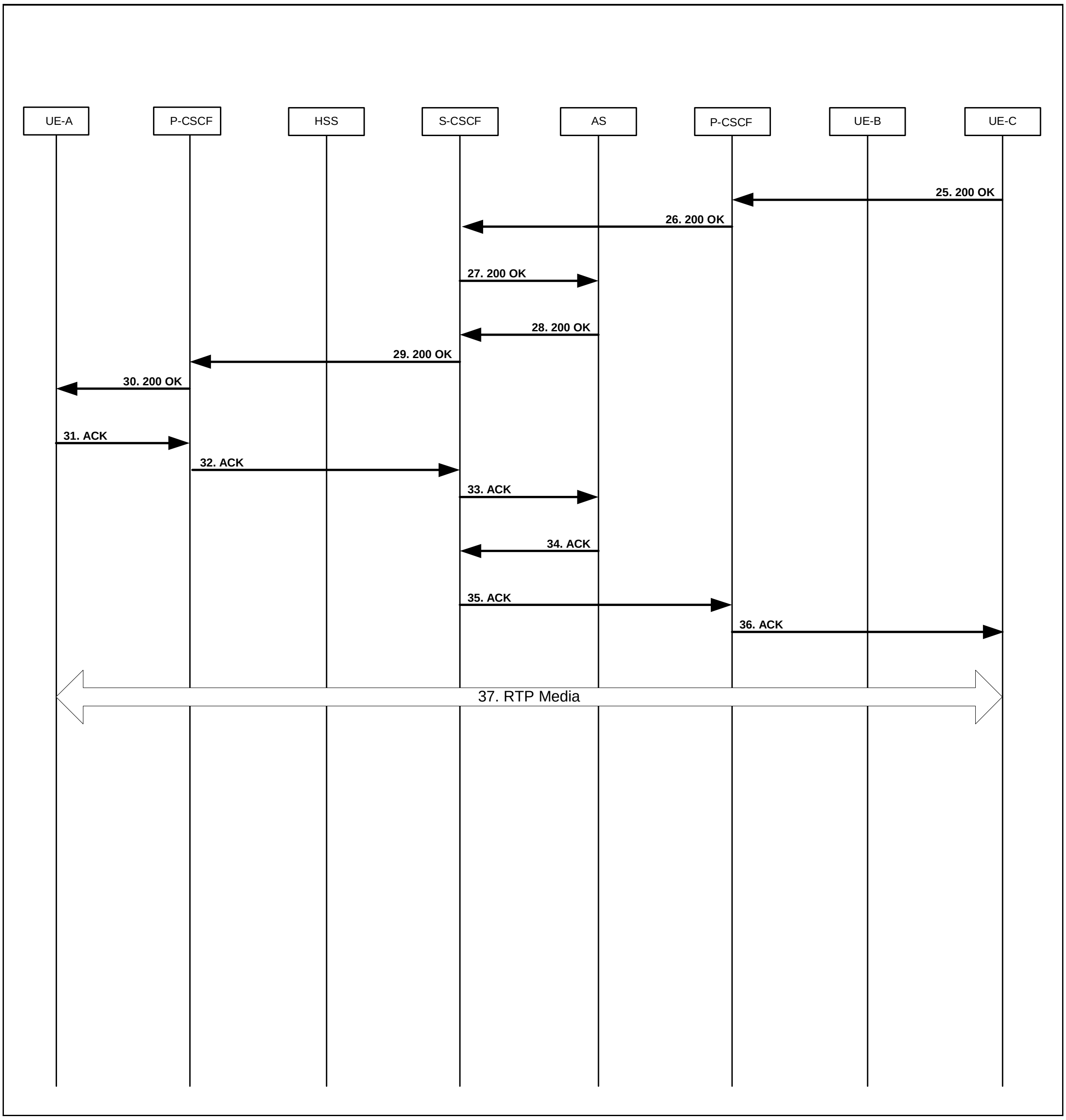


UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
25. 200 OK
26. 200 OK
27. 200 OK
28. 200 OK
29. 200 OK
30. 200 OK
31. ACK
32. ACK
33. ACK
34. ACK
35. ACK
36. ACK
37. RTP Media

SSS__XXSSCD 02	**NGN reference to: [17]**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CD- Immediate response.	
Configuration	The user A and the user C are in network N1. The user B is in network N2 and is provided with CD- Immediate response ("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).	
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 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).	
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:		

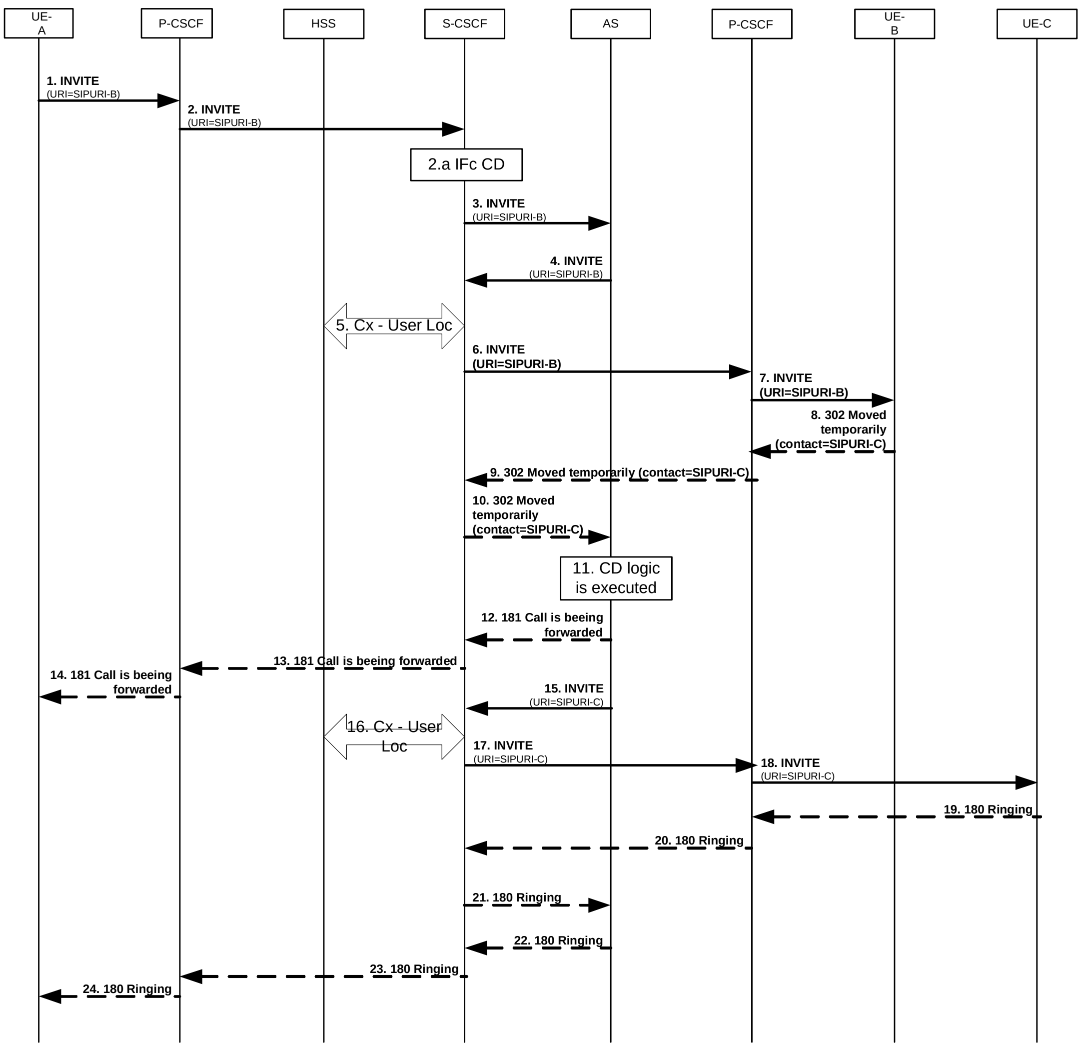


Continued

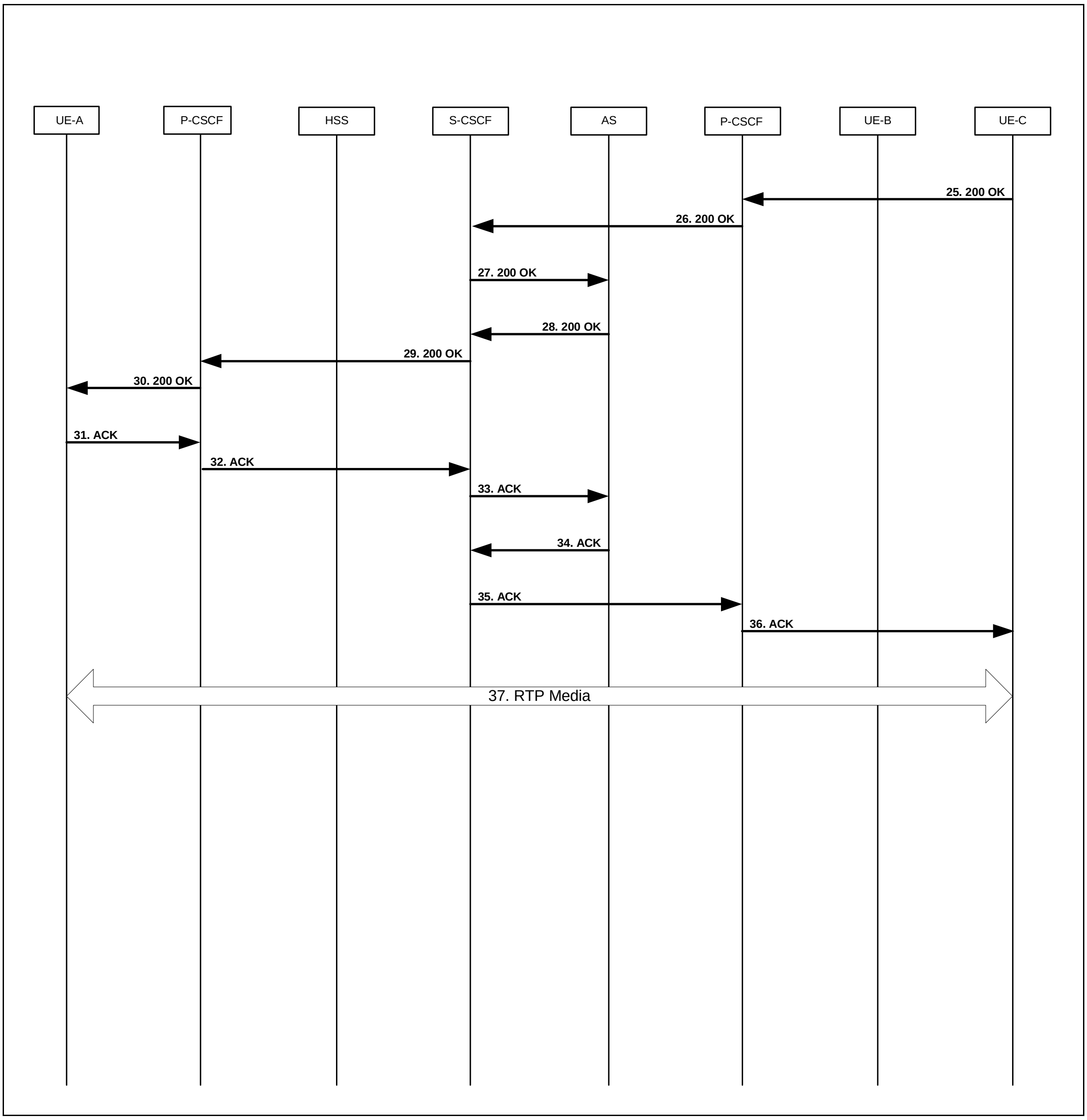

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
25. 200 OK
26. 200 OK
27. 200 OK
28. 200 OK
29. 200 OK
30. 200 OK
31. ACK
32. ACK
33. ACK
34. ACK
35. ACK
36. ACK
37. RTP Media

SSS__XXSSCD 03	NGN reference to: [17]
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CD- Immediate response.
Configuration	The user A and the user C are in network N1. The user B is in network N2 and is provided with CD- Immediate response ("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).
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 presentation not allowed – TIR) and user C is not informed of the forwarding number (user B has presentation not allowed).
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:



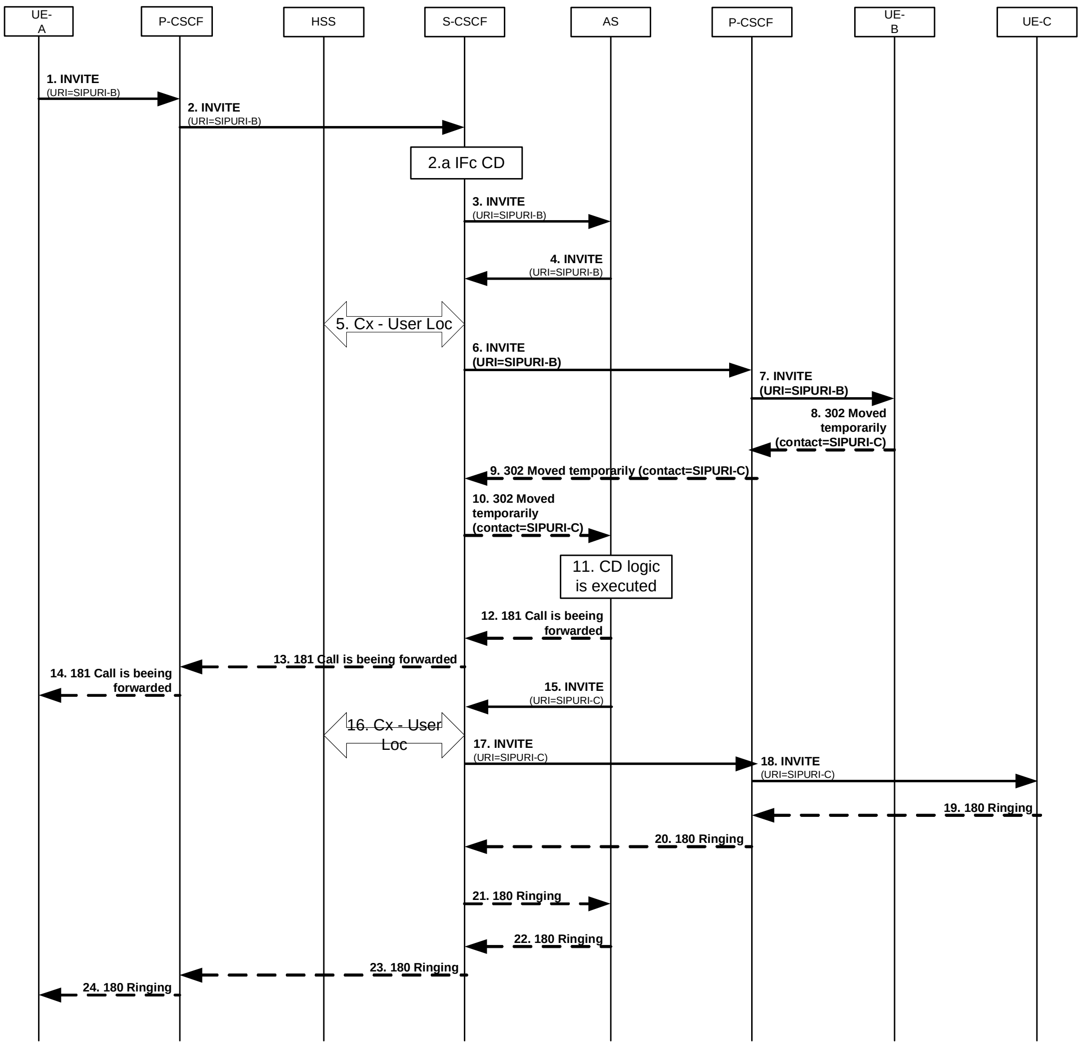
Continued



SSS__XXSSCD 04	NGN reference to: [17]	
TSS reference:	SIP-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 N2 and is provided with CD- Immediate response. 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).	
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 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).	

SSS__XXSSCD 04	NGN reference to: [17]	
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:



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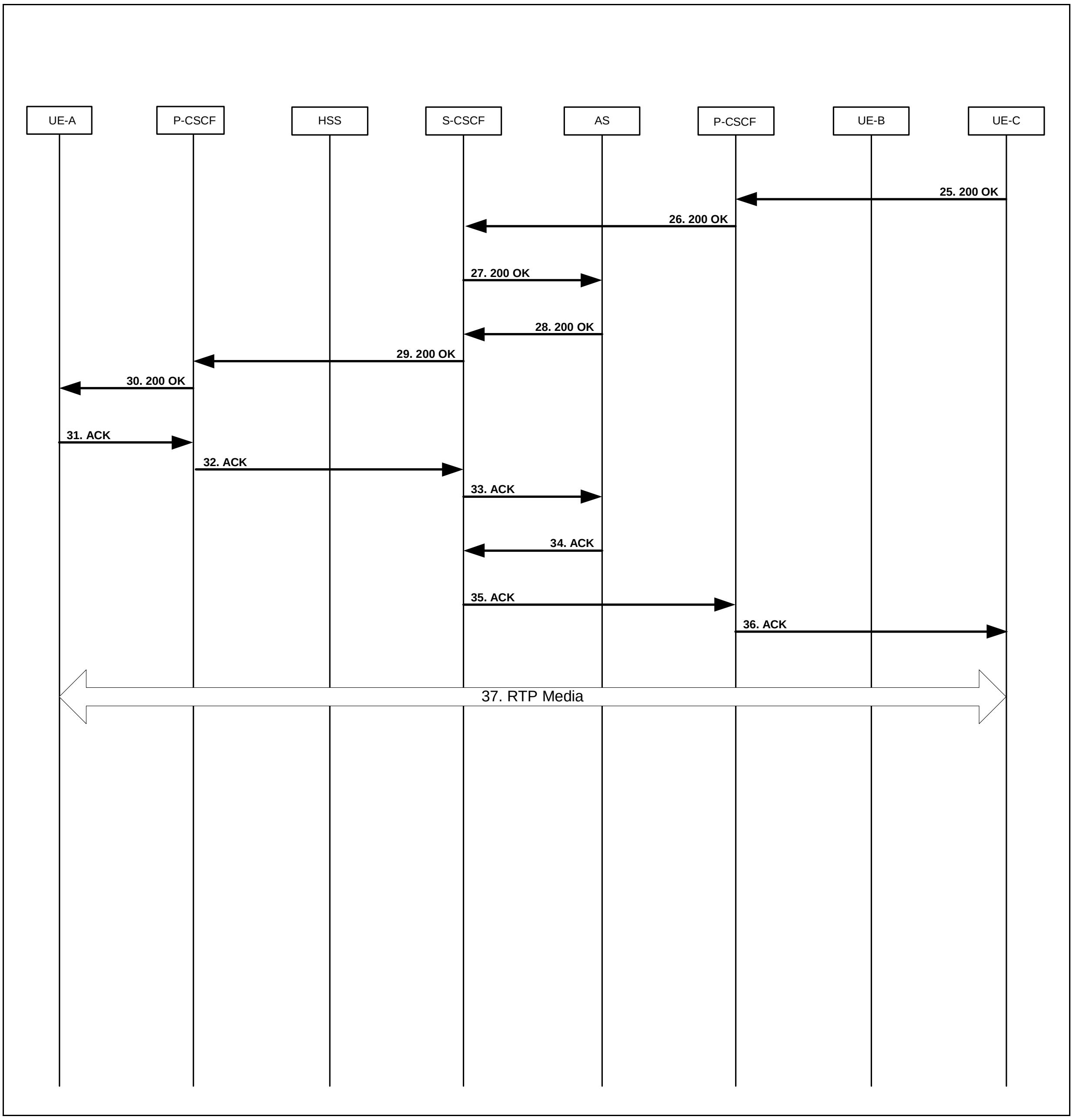
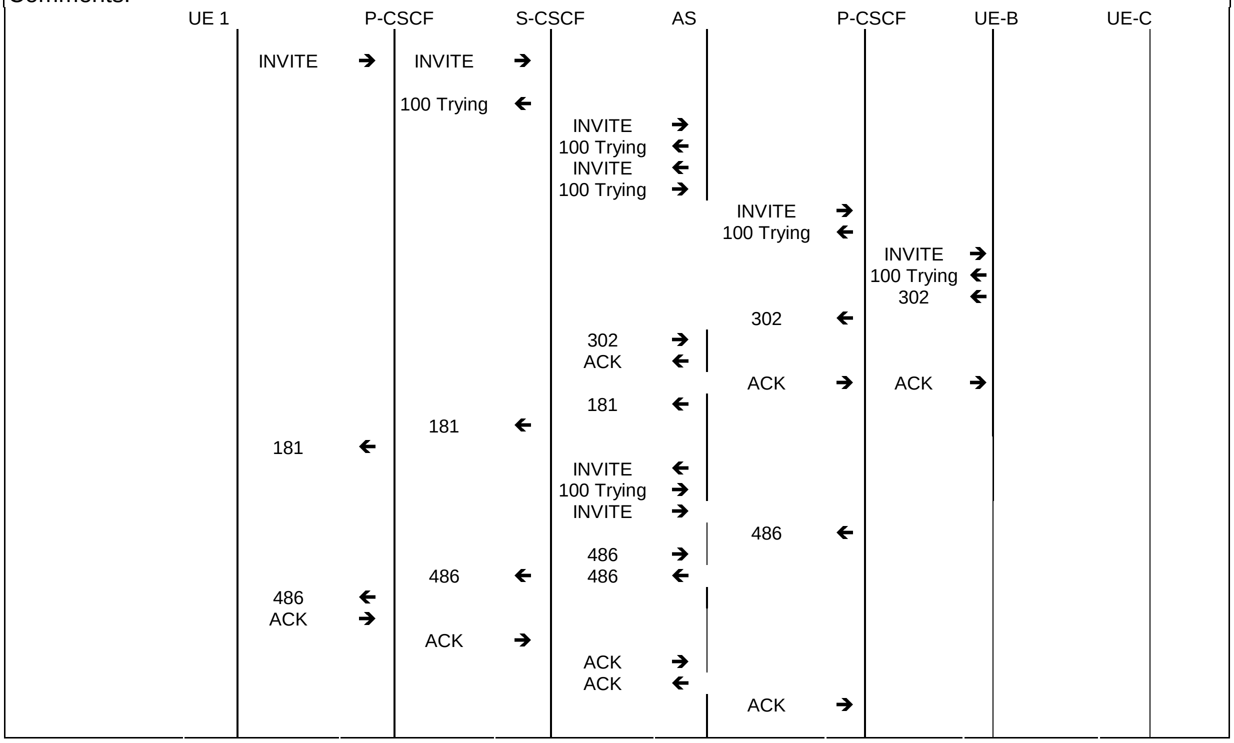

UE-A
P-CSCF
HSS
S-CSCF
AS
P-CSCF
UE-B
UE-C
25. 200 OK
26. 200 OK
27. 200 OK
28. 200 OK
29. 200 OK
30. 200 OK
31. ACK
32. ACK
33. ACK
34. ACK
35. ACK
36. ACK
37. RTP Media

SSS__XXSSCD 05	**NGN reference to: [17]**
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CD- Immediate response
Configuration	The user B is in network N2 and is provided with CD- Immediate response
Selection criteria:	• Call forwarding uncontitional supported. • user C is network determined user busy. • User B has activated the CALL DEFLECTION service.
Test purpose:	To verify that a call is released correctly if CALL DEFLECTION 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:	

	UE 1	I-CSCF	S-CSCF	AS	P-CSCF	UE-B	UE-C
	INVITE →	INVITE →					
		100 Trying ←					
			INVITE →				
			100 Trying ←				
			INVITE ←				
			100 Trying →				
				INVITE →			
				100 Trying ←			
					INVITE →		
					100 Trying ←		
					302 ←		
				302 ←			
			302 →				
			ACK ←				
				ACK →	ACK →		
			181 ←				
		181 ←					
	181 ←						
			INVITE ←				
			100 Trying →				
			INVITE →			INVITE →	
				486 ←		486 ←	
			486 →			ACK →	
		486 ←	486 ←				
	486 ←						
	ACK →						
		ACK →					
			ACK →				
			ACK ←				
				ACK →			

SSS__XXSSCD 06	**NGN reference to: [17]**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CD- Immediate response CD- Immediate response.	
Configuration	The user B is in network N2 and is provided with CD- Immediate response.	
Selection criteria:	• Call forwarding unconditional supported. • user C is network determined network busy. • User B has activated the CALL DEFLECTION service.	
Test purpose:	To verify that a call is released correctly if CALL DEFLECTION 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:



6.5.2.11 CONF

6.5.2.11.1 Conference creation

SSS__XXSSCONF_ CRE_001	**NGN reference to: [20] clause 5.3.1.3.3**	
TSS reference:	SIP-SIP-SIP/Supplementary_servicesCONF	
Configuration	CONF	
Selection criteria:	CONF	
Test purpose:	Conference creation by Three-way session creation. REFER request to the user, Conference event package is 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 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. 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 SUBSCRIBE: Request URI contained the conference URI header contains "conference" 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** application/conference-info+xml contains connected, dialled-in 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** application/conference-info+xml contains connected, dialled-in	
Continued		

Comments:						
SIP#1		**Focus**		**SIP#2**		**SIP#3**
Establishment of session #1						
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
200 OK (INVITE)	←		←	200 OK (INVITE)		
ACK	→		→	ACK		
INVITE (sendonly)	→		→	INVITE (sendonly)		
200 OK (recvonly)	←		←	200 OK (recvonly)		
ACK	→		→	ACK		
	Establishment of session #2					
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	←				←	200 OK (INVITE)
ACK	→				→	ACK
Conference creation						
INVITE	→	INVITE				
200 OK	←	200 OK				
ACK	→	ACK				
SUBSCRIBE	→	SUBSCRIBE				
200 OK	←	200 OK				
NOTIFY	←	NOTIFY				
200 OK NOTIFY	→	200 OK NOTIFY				
Inviting SIP#2 to the conference						
REFER	→		→	REFER		
202 Accepted	←		←	202 Accepted		
		INVITE	←	INVITE		
NOTIFY	←		←	NOTIFY 1		
200 OK NOTIFY	→		→	200 OK NOTIFY		
		200 OK	→	200 OK		
		ACK	←	ACK		
NOTIFY	←		←	NOTIFY 2		
200 OK NOTIFY	→		→	200 OK NOTIFY		
BYE	→		→	BYE		
200 OK (BYE)	←		←	200 OK (BYE)		
NOTIFY	←	NOTIFY 3				
200 OK NOTIFY	→	200 OK NOTIFY				
	Inviting SIP#3 to the conference					
REFER	→				→	REFER
202 Accepted	←				←	202 Accepted
		INVITE	←		←	INVITE
NOTIFY	←				←	NOTIFY 4
200 OK NOTIFY	→				→	200 OK NOTIFY
		200 OK	→		→	200 OK
		ACK	←		←	ACK
NOTIFY	←				←	NOTIFY 5
200 OK NOTIFY	→				→	200 OK NOTIFY
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)
NOTIFY	←	NOTIFY 6				
200 OK	→	200 OK				

SSS__XXSSCONF_CRE_002	**NGN reference to: [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
Establishment of session #1						
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
200 OK (INVITE)	←		←	200 OK (INVITE)		
ACK	→		→	ACK		
INVITE (sendonly)	→		→	INVITE (sendonly)		
200 OK (recvonly)	←		←	200 OK (recvonly)		
ACK	→		→	ACK		
	Establishment of session #2					
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	←				←	200 OK (INVITE)
ACK	→				→	ACK
Conference creation						
INVITE	→	INVITE				
200 OK	←	200 OK				
ACK	→	ACK				
Inviting SIP#2 to the conference						
REFER	→		→	REFER		
202 Accepted	←		←	202 Accepted		
		INVITE	←	INVITE		
NOTIFY	←		←	NOTIFY 1		
200 OK NOTIFY	→		→	200 OK NOTIFY		
		200 OK	→	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 conference					
REFER	→				→	REFER
202 Accepted	←				←	202 Accepted
		INVITE	←		←	INVITE
NOTIFY	←				←	NOTIFY 3
200 OK NOTIFY	→				→	200 OK NOTIFY
		200 OK	→		→	200 OK
		ACK	←		←	ACK
NOTIFY	←				←	NOTIFY 4
200 OK NOTIFY	→				→	200 OK NOTIFY
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)

SSS__XXSSCONF_CRE_003	**NGN reference to: [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 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 in the Contact header field. 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** application/conference-info+xml contains connected, dialled-out.	
Continued		

Comments:

SIP#1		Focus		SIP#2		SIP#3
Establishment of session #1						
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
200 OK (INVITE)	←		←	200 OK (INVITE)		
ACK	→		→	ACK		
INVITE (sendonly)	→		→	INVITE (sendonly)		
200 OK (recvonly)	←		←	200 OK (recvonly)		
ACK	→		→	ACK		
		Establishment of session #2				
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	←				←	200 OK (INVITE)
ACK	→				→	ACK
Conference creation						
INVITE	→	INVITE				
200 OK	←	200 OK				
ACK	→	ACK				
SUBSCRIBE	→	SUBSCRIBE				
200 OK	←	200 OK				
Inviting SIP#2 to the conference						
REFER 1	→	REFER				
202 Accepted	←	202 Accepted				
		INVITE	→	INVITE		
NOTIFY	←	NOTIFY 1				
200 OK NOTIFY	→	200 OK NOTIFY				
		200 OK	←	200 OK		
		ACK	→	ACK		
NOTIFY	←	NOTIFY 2				
200 OK NOTIFY	→	200 OK NOTIFY				
BYE	→		→	BYE		
200 OK (BYE)	←		←	200 OK (BYE)		
NOTIFY	←	NOTIFY 3				
200 OK NOTIFY	→	200 OK NOTIFY				
		Inviting SIP#3 to the conference				
REFER 2	→	REFER				
202 Accepted	←	202 Accepted				
		INVITE	→		→	INVITE
NOTIFY	←	NOTIFY 4				
200 OK NOTIFY	→	200 OK NOTIFY				
		200 OK	←		←	200 OK
		ACK	→		→	ACK
NOTIFY	←	NOTIFY 5				
200 OK NOTIFY	→	200 OK NOTIFY				
BYE	→				→	BYE
200 OK (BYE)	←				←	200 OK (BYE)
NOTIFY	←	NOTIFY 6				
200 OK	→	200 OK				

SSS__XXSSCONF_CRE_004	**NGN reference to: [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 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.**	

Comments:							
SIP#1		**Focus**			**SIP#2**		**SIP#3**
Establishment of session #1							
INVITE	→			→	INVITE		
180 Ringing	←			←	180 Ringing		
200 OK (INVITE)	←			←	200 OK (INVITE)		
ACK	→			→	ACK		
INVITE (sendonly)	→			→	INVITE (sendonly)		
200 OK (recvonly)	←			←	200 OK (recvonly)		
ACK	→			→	ACK		
		Establishment of session #2					
INVITE	→					→	INVITE
180 Ringing	←					←	180 Ringing
200 OK (INVITE)	←					←	200 OK (INVITE)
ACK	→					→	ACK
Conference creation							
INVITE	→	INVITE					
200 OK	←	200 OK					
ACK	→	ACK					
Inviting SIP#2 to the conference							
REFER 1	→	REFER					
202 Accepted	←	202 Accepted					
			INVITE	→	INVITE		
NOTIFY	←	NOTIFY1					
200 OK NOTIFY	→	200 OK NOTIFY					
			200 OK	←	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 conference					
REFER 2	→	REFER					
202 Accepted	←	202 Accepted					
			INVITE	→		→	INVITE
NOTIFY	←	NOTIFY 3					
200 OK NOTIFY	→	200 OK NOTIFY					
			200 OK	←		←	200 OK
			ACK	→		→	ACK
NOTIFY	←	NOTIFY 4					
200 OK NOTIFY	→	200 OK NOTIFY					
BYE	→					→	BYE
200 OK (BYE)	←					←	200 OK (BYE)

SSS__XXSSCONF_CRE_005	**NGN reference to: [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, 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 URI?Replaces=call-id1%3Bto-tagsession1%3Bfrom-tagSession1; method=INVITE>. 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-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 #1						
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
200 OK (INVITE)	←		←	200 OK (INVITE)		
ACK	→		→	ACK		
INVITE (sendonly)	→		→	INVITE (sendonly)		
200 OK (recvonly)	←		←	200 OK (recvonly)		
ACK	→		→	ACK		
		Establishment of session #2				
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	←				←	200 OK (INVITE)
ACK	→				→	ACK
Conference creation						
INVITE	→	INVITE				
200 OK	←	200 OK				
ACK	→	ACK				
Inviting SIP#2 to the conference						
REFER 1	→		→	REFER		
202 Accepted	←		←	202 Accepted		
		INVITE	←	INVITE		
NOTIFY	←		←	NOTIFY		
200 OK NOTIFY	→		→	200 OK NOTIFY		
		200 OK	→	200 OK		
		ACK	←	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
		200 OK	→		→	200 OK
		ACK	←		←	ACK
NOTIFY	←				←	NOTIFY
200 OK NOTIFY	→				→	200 OK NOTIFY
BYE	←				←	BYE 2
200 OK (BYE)	→				→	200 OK (BYE)

SSS__XXSSCONF_CRE_06	**NGN reference to:** **[20] clause 4.5.2.1.2**	
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 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 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-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-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:

SIP#1		Focus		SIP#2		SIP#3
Establishment of session #1						
INVITE	→		→	INVITE		
180 Ringing	←		←	180 Ringing		
200 OK (INVITE)	←		←	200 OK (INVITE)		
ACK	→		→	ACK		
INVITE (sendonly)	→		→	INVITE (sendonly)		
200 OK (recvonly)	←		←	200 OK (recvonly)		
ACK	→		→	ACK		
	Establishment of session #2					
INVITE	→				→	INVITE
180 Ringing	←				←	180 Ringing
200 OK (INVITE)	←				←	200 OK (INVITE)
ACK	→				→	ACK
Conference creation						
INVITE	→	INVITE				
200 OK	←	200 OK				
ACK	→	ACK				
Inviting SIP#2 to the conference						
REFER 1	→	REFER				
202 Accepted	←	202 Accepted				
		INVITE 4	→	INVITE		
		200 OK	←	200 OK		
		ACK	→	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 5	→		→	INVITE
		200 OK	←		←	200 OK
		ACK	→		→	ACK
NOTIFY	←	NOTIFY				
200 OK NOTIFY	→	200 OK NOTIFY				
BYE	←				←	BYE 2
200 OK (BYE)	→				→	200 OK (BYE)

SSS__XXSSCONF_CRE_07	**NGN reference to:** **[20] clause 5.3.1.3.2**	
TSS reference:	SIP-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. 183 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** **Focus** INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE → 200 OK UPDATE ← 200 OK (INVITE) ← ACK →		

SSS__XXSSCONF_CRE_08	NGN reference to: [20] clause 5.3.1.3.3	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	CONF	
Selection criteria:	Is 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 temporary 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. 183 Temporary 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		Focus
INVITE	→	
183 Session Progress	←	
PRACK	→	
200 OK PRACK	←	
UPDATE	→	
200 OK UPDATE	←	
200 OK (INVITE)	←	
ACK	→	

SSS__XXSSCONF_CRE_09	NGN reference to: [20] clause 5.3.2.3.1	
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 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:		

SIP#1		Focus
INVITE	→	
488 Not Acceptable Here	←	
ACK	→	

SSS__XXSSCONF_CRE_10	**NGN reference to: [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: **SIP#1** **Focus** INVITE → 488 Not Acceptable Here ← ACK →		

6.5.2.11.2 Joining a conference

SSS__XXSSCONF_ Join _01	**NGN reference to: [20] clause 5.3.2.4**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	CONF	
Selection criteria:	CONF	
Test purpose:	*Participant dial-in the conference, the conference URI is used. The Participant subscribes the conference event package.* SIP#1 established a conference. SIP#2 joins in that conference. An INVITE request is received at the conferencing AS the conference URI is known at the SIP#2. SIP#2 subscribes 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. INVITE 2: Request URI contained the conference URI. 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, Event contains "conference". Event header contains "conference". NOTIFY: Event contains conference; Subscription-State contains active; expires=XXXX. application/conference-info+xml contains connected, dialled-in.	

Comments:

SIP#1		**Focus**		**SIP#2**
Conference creation				
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
SIP#2 joining in the conference				
		INVITE	←	INVITE 2
		18x	→	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		

SSS__XXSSCONF_ Joint _02	**NGN reference to: [20] clause 5.3.2.4**	
TSS reference:	SIP-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		**Focus**		**SIP#2**
Conference creation				
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
SIP#2 joining in the conference				
		INVITE	←	INVITE 2
		18x	→	18x
		200 OK INVITE	→	200 OK INVITE
		ACK	←	ACK

SSS__XXSSCONF_ Join_003	**NGN reference to: [20] clause 5.3.2.4**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF	
Configuration	CONF	
Selection criteria:	CONF	
Test purpose:	*Participant dial-in the conference, the conference URI is not allocated, the request is rejected.* SIP#1 established a conference. SIP#2 joins in that conference. The conference URI in the INVITE request is not allocated at the focus. The request is rejected with the final response 488 Not Acceptable Here.	
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 not allocated in the focus	

Comments:

SIP#1		**Focus**		**SIP#2**
Conference creation				
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
SIP#2 joining in the conference				
		INVITE	←	INVITE 2
		488 Not Acceptable Here	→	488 Not Acceptable Here
		ACK	←	ACK

6.5.2.11.3 Inviting other users to a conference

SSS__XXSSCONF_INV_001	**NGN reference to: [20] clause 5.3.2.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF	
Configuration	CONF	
Selection criteria:	CONF	
Test purpose:	*Inviting participant by sending REFER to the focus. The conference event package is subscribed.* SIP#1 established a conference. SIP#1 invites SIP#2 to join into the conference. The SIP#1 sends a REFER to the focus; the focus sends an INVITE request to SIP#2 to invite the SIP#2 to the conference. 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: Request URI contained the **conference 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 conference URI. The P-Asserted-Identity contains the conference URI. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of **SIP#1.** 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-out. NOTIFY 4 Event contains **conference**; Subscription-State contains **active** application/conference-info+xml contains connected, dialled-out.	
Continued		

Comments:

SIP#1		**Focus**		**SIP#2**
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	←	NOTIFY 4		
200 OK NOTIFY	→	200 OK NOTIFY		

SSS__XXSSCONF_INV_002	**NGN reference to: [20] clause 5.3.2.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	CONF	
Selection criteria:	CONF	
Test purpose:	*Inviting participant by sending REFER to the focus. The conference event package is not subscribed.* SIP#1 established a conference. SIP#1 invites SIP#2 to join into the conference. The SIP#1 sends a REFER to the focus; the focus sends an INVITE request to SIP#2 to invite the SIP#2 to the conference. 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. **SIP header values:** REFER: Request URI contained the conference URI. Refer-To contains the URI of SIP#2, method=invite. Referred-By contains SIP URI of SIP#1. INVITE 2: Request URI contained the conference URI. The P-Asserted-Identity contains the conference URI. "isfocus" feature parameter indicated in Contact header field conference URI contained in the Contact header field. Referred-By contains SIP or tel URI of SIP#1. 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.	

Comments:

SIP#1		**Focus**		**SIP#2**
Conference creation				
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
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		

SSS__XXSSCONF_ INV_003	**NGN reference to: [20] clause 5.3.2.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	CONF	
Selection criteria:	CONF	
Test purpose:	*Inviting participant by sending REFER to the participant. The conference event package is subscribed.* SIP#1 established a conference. SIP#1 invites SIP#2 to join into the conference by sending a REFER to this participant. The participant sends an INVITE request to the focus to dial in. 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: Request URI contained the URI of **SIP#2.** Refer-To contains the **conference URI**, method=invite. Referred-By contains the URI of **SIP#1.** 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. Referred-By contains SIP or tel URI of **SIP#1.** NOTIFY 1 Event contains **conference**; Subscription-State contains **active.** Event contains conference; Subscription-State contains active message/sipfrag contains **SIP/2.0 100 Trying.** NOTIFY 2 Event contains **conference**; Subscription-State contains **active.** 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.	
Continued		

Comments:

SIP#1		**Focus**		**SIP#2**
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
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 NOTIFY	→		→	200 OK NOTIFY
			←	SUBSCRIBE
			→	200 OK (SUBSCRIBE)
		NOTIFY 3	→	NOTIFY
		200 OK NOTIFY	←	200 OK NOTIFY
NOTIFY	←	NOTIFY 4		
200 OK NOTIFY	→	200 OK NOTIFY		

SSS__XXSSCONF_INV_004	**NGN reference to: [20] clause 5.3.2.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	CONF	
Selection criteria:	CONF	
Test purpose:	*Inviting participant by sending REFER to the participant. The conference event package is not subscribed.* SIP#1 established a conference. SIP#1 invites SIP#2 to join into the conference by sending a REFER to this participant. The participant sends an INVITE request to the focus to dial in. 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. **SIP header values:** REFER: Request URI contained the URI of **SIP#2.** Refer-To contains the **conference URI**, method=invite. Referred-By contains the URI of **SIP#1.** INVITE 2: Request URI contained 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. Referred-By contains SIP or tel URI of **SIP#1.** 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.**	

Comments:

SIP#1		**Focus**		**SIP#2**
Conference creation				
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
		INVITE	←	INVITE 2
		180 Ringing	→	180 Ringing
		200 OK INVITE	→	200 OK INVITE
		ACK	←	ACK
NOTIFY	←	NOTIFY 2		
200 OK NOTIFY	→	200 OK NOTIFY		

SSS__XXSSCONF_INV_005	**NGN reference to: [20] clause 5.3.2.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	CONF	
Selection criteria:	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 focus to 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; 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 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	→	INVITE			
200 OK (INVITE)	←	200 OK (INVITE)			
ACK	→	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
			200 OK NOTIFY	→	200 OK NOTIFY
				←	SUBSCRIBE
				→	200 OK (SUBSCRIBE)
			NOTIFY 3	→	NOTIFY
			200 OK NOTIFY	←	200 OK NOTIFY
NOTIFY	←	NOTIFY 4			
200 OK NOTIFY	→	200 OK NOTIFY			

SSS__XXSSCONF_ INV_00	**NGN reference to:** **[20] clause 5.3.2.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	CONF	
Selection criteria:	CONF	
Test purpose:	Focus invites 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 focus to dial in. 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: 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 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**	

Comments:

SIP#1		**Focus**		**SIP#2**
Conference creation				
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	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
		200 OK NOTIFY	→	200 OK NOTIFY

6.5.2.11.4 Leaving a conference

SSS__XXSSCONF _LEAV_001	**NGN reference to: [20] clause 5.3.2.6**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF	
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 to the basic call procedures. 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:** NOTIFY 1 Event contains **conference**; Subscription-State contains **terminated** xml document: **<user>** uri=SIP#2, status is **departed.** NOTIFY 2 Event contains **conference**; Subscription-State contains **active** xml document: **<user>** uri=SIP#2, status is **departed.**	

Comments:

SIP#1		**Focus**		**SIP#2**
Conference creation				
INVITE	→	INVITE		
200 OK (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 communicaton				
Participant leaves the converence				
			←	BYE
			→	200 OK BYE
		NOTIFY 1	→	NOTIFY
		200 NOTIFY	←	200 NOTIFY
NOTIFY	←	NOTIFY 2		
200 NOTIFY	→	200 NOTIFY		

SSS__XXSSCONF _LEAV_002	**NGN reference to: [20] clause 5.3.2.6**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	CONF	
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		
200 OK (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 communicaton				
Participant leaves the converence				
			←	BYE
			→	200 OK BYE

SSS__XXSSCONF _LEAV_003	**NGN reference to: [20] clause 5.3.2.6**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	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 request from SIP#1 contains the conference URI and the method value BYE in the Refer-To header field. 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>** uri=SIP#2, status is **departed.** NOTIFY 4 Event contains **conference**; Subscription-State contains **active** xml document: **<user>** uri=SIP#2, status is **departed.**	

Comments:

SIP#1		**Focus**		**SIP#2**
Conference creation				
INVITE	→	INVITE		
200 OK (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 communicaton				
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
			←	BYE
			→	200 OK BYE
NOTIFY	←		←	NOTIFY 2
200 OK NOTIFY	→		→	200 OK NOTIFY
		NOTIFY 3	→	NOTIFY
		200 NOTIFY	←	200 NOTIFY
NOTIFY	←	NOTIFY 4		
200 NOTIFY	→	200 NOTIFY		

<table>
<tr><td>SSS__XXSSCONF_ LEAV_004</td><td>NGN reference to:
[20] clause 5.3.2.6</td><td></td></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-SIP-SIP/Supplementary_services/ CONF</td></tr>
<tr><td>Configuration</td><td colspan="2">CONF</td></tr>
<tr><td>Selection criteria:</td><td colspan="2">CONF</td></tr>
<tr><td>Test purpose:</td><td colspan="2">A participant wishes to remove an other participant from the conference. The conference event package is not subscribed.

SIP#1 wishes to remove SIP#2 from the conference. SIP#2 receives a REFER request from SIP#1 contains the conference URI and the method value BYE in the Refer-To header field. SIP#2 sends a BYE to the focus. The conference event package is not subscribed.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">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.</td></tr>
<tr><td colspan="3">Comments:</td></tr>
</table>

SIP#1		Focus		SIP#2
Conference creation				
INVITE	→	INVITE		
200 OK (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 communicaton				
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
			←	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

SSS__XXSSCONF_ REMOV_001	**NGN reference to: [20] clause 5.3.2.6**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
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.* SIP#2 asks the focus by sending a REFER request 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.	
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 **conference URI.** Refer-To contains the **SIP#2 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.** 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 **terminated** xml document: **<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.**	

Comments:

SIP#1		**Focus**		**SIP#2**
Conference creation				
INVITE	→	INVITE		
200 OK (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 communicaton		
		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		
		focus removes SIP#2 from the conference		
			→	BYE
			←	200 OK BYE
NOTIFY	←	NOTIFY 2		
200 NOTIFY	→	200 NOTIFY		
		NOTIFY 3	→	NOTIFY
		200 NOTIFY	←	200 NOTIFY
NOTIFY	←	NOTIFY 4		
200 NOTIFY	→	200 NOTIFY		

SSS__XXSSCONF_ REMOV_002	**NGN reference to: [20] clause 5.3.2.5**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	CONF	
Selection criteria:	CONF	
Test purpose:	*A participant asks the focus to remove a participant from the conference. The conference event package is not subscribed.* SIP#2 asks the focus by sending a REFER request 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 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. **SIP header values:** REFER 1: Request URI contained the URI of **conference URI.** Refer-To contains the **SIP#2 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.** NOTIFY 2 Event contains **conference**; Subscription-State contains **active** message/sipfrag contains **SIP/2.0 200 OK.**	

Comments:

SIP#1		**Focus**		**SIP#2**
		Conference creation		
INVITE	→	INVITE		
200 OK (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 communicaton		
		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		
		focus removes SIP#2 from the conference		
			→	BYE
			←	200 OK BYE
NOTIFY	←	NOTIFY 2		
200 NOTIFY	→	200 NOTIFY		

SSS__XXSSCONF_ REMOV_003	**NGN reference to: [20] clause 5.3.2.6**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
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.	
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:** NOTIFY 1 Event contains **conference**; Subscription-State contains **terminated** xml document: **<user>** uri=SIP#2, status is **departed.** NOTIFY 2 Event contains **conference**; Subscription-State contains **active** xml document: **<user>** uri=SIP#1, status is **departed.**	

Comments:

SIP#1		**Focus**		**SIP#2**
Conference creation				
INVITE	→	INVITE		
200 OK (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 communicaton		
		focus removes SIP#2 from the conference		
			→	BYE
			←	200 OK BYE
		NOTIFY 1	→	NOTIFY
		200 NOTIFY	←	200 NOTIFY
NOTIFY	←	NOTIFY 2		
200 NOTIFY	→	200 NOTIFY		

SSS__XXSSCONF_ REMOV_004	**NGN reference to: [20] clause 5.3.2.6**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	CONF	
Selection criteria:	CONF	
Test purpose:	*The focus removes a participant from the conference. The conference event package is not 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 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		
200 OK (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 communicaton		
		focus removes SIP#2 from the conference		
			→	BYE
			←	200 OK BYE

SSS__XXSSCONF_ REMOV_005	**NGN reference to: [20] clause 5.3.2.6**	
TSS reference:	SIP-SIP-SIP/Supplementary_services/ CONF.	
Configuration	CONF	
Selection criteria:	CONF	
Test purpose:	*The conference owner releases the entire conference by sending a BYE to the focus. The conference event package ist subscribed.* SIP#1, the conference owner, sends a BYE request to the focus. The entire conference is released. 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:** BYE: Request URI = conference URI. NOTIFY 1 Event contains **conference**; Subscription-State contains **terminated** xml document: **<user>** uri=SIP#2, status is **departed.** NOTIFY 2 Event contains **conference**; Subscription-State contains **terminated** xml document: **<user>** uri=SIP#1, status is **departed.**	

Comments:

SIP#1		**Focus**		**SIP#2**
		Conference creation		
INVITE	→	INVITE		
200 OK (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 communicaton		
		SIP#1 wishes to finish the entire conference		
BYE	→	BYE		
200 OK BYE	←	200 OK BYE		
		focus removes SIP#2 from the conference		
			→	BYE
			←	200 OK BYE
		NOTIFY 1	→	NOTIFY
		200 NOTIFY	←	200 NOTIFY
NOTIFY	←	NOTIFY 2		
200 NOTIFY	→	200 NOTIFY		

SSS__XXSSCONF_REMOV_006	**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 owner releases the entire conference by sending a BYE to the focus. The conference event package is not subscribed.* SIP#1, the conference owner, sends a BYE request to the focus. The entire conference is released. 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. **SIP header values:** BYE: Request URI = conference URI	

Comments:

SIP#1		**Focus**		**SIP#2**
		Conference creation		
INVITE	→	INVITE		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
SIP#2 joining in the conference				
		INVITE	←	INVITE 2
		180 Ringing	→	180 Ringing
		200 OK INVITE	→	200 OK INVITE
		ACK	←	ACK
		Conference communicaton		
		SIP#1 wishes to finish the entire conference		
BYE	→	BYE		
200 OK BYE	←	200 OK BYE		
		focus removes SIP#2 from the conference		
			→	BYE
			←	200 OK BYE

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 focus. The conference event package ist subscribed.* Thehe conference policy decides to terminate the entire conference by sending a BYE request to each participant. 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:** NOTIFY 1 Event contains **conference**; Subscription-State contains **terminated** xml document: **<user>** uri=SIP#2, status is **departed.** NOTIFY 2 Event contains **conference**; Subscription-State contains **terminated** xml document: **<user>** uri=SIP#1, status is **departed.**	

Comments:

SIP#1		**Focus**		**SIP#2**
Conference creation				
INVITE	→	INVITE		
200 OK (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 communicaton		
		focus removes SIP#1 from the conference		
BYE	←	BYE		
200 OK BYE	→	200 OK BYE		
		focus removes SIP#2 from the conference		
			→	BYE
			←	200 OK BYE
		NOTIFY 1	→	NOTIFY
		200 NOTIFY	←	200 NOTIFY
NOTIFY	←	NOTIFY 1		
200 NOTIFY	→	200 NOTIFY		

SSS__XXSSCONF_ REMOV_008	**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 focus. The conference event package ist not subscribed.* The conference policy decides to terminate the entire conference by sending a BYE request to each participant. 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		
200 OK (INVITE)	←	200 OK (INVITE)		
ACK	→	ACK		
		SIP#2 joining in the conference		
		INVITE	←	INVITE 2
		180 Ringing	→	180 Ringing
		200 OK INVITE	→	200 OK INVITE
		ACK	←	ACK
		Conference communicaton		
		focus removes SIP#1 from the conference		
BYE	←	BYE		
200 OK BYE	→	200 OK BYE		
		focus removes SIP#2 from the conference		
			→	BYE
			←	200 OK BYE

Annex A (informative): Bibliography

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- IETF RFC 3325 (2002): "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks".
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- draft-ietf-sipping-basic-call-flows-02.txt: "Session Initiation Protocol Basic Call Flow Examples".

History

Document history		
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