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Technical Specification

**IMS Network Testing (INT);
NGN/IMS interconnection tests at the Ic Interface;
Part 1: Protocol Implementation Conformance
Statement (PICS)**

Reference

DTS/INT-00058

Keywords

SIP, testing, UNI, interconnection

ETSI

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

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

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
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Foreword

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

1 Scope

The present document specifies the Test Suite Structure and Test Purposes (TSS&TP) for **NGN/IMS interconnection tests at the Ic Interface** to verify the overall compatibility of SIP, ISDN and non-ISDN (PSTN) over the national or international NGN networks under consideration of the use of End Devices in the relevant networks (recommended by the network operator). The TSS&TP specification covers the procedures described in TS 124 229 [2] and TS 129 165 [1] respectively.

The specified Test Purposes are the basis for bilateral tests between national or international network operators. Even if tests between network operators is agreed, exactly the test purposes defined in the current document are be performed. Modification of the requirements as described in TS 124 229 [2] and TS 129 165 [1] based on national requirements needs additional Test Purposes not described in the present document. This additional test may be defined and agreed between the test staff of the network operators.

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the reference document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

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2.1 Normative references

The following referenced documents are necessary for the application of the present document.

[1] ETSI TS 129 165: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Inter-IMS Network to Network Interface (NNI) (3GPP TS 29.165 Release 10)".

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

[3] IETF RFC 4566 (2006): "SDP: Session Description Protocol".

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

[5] IETF RFC 3264 (2002): "An Offer/Answer Model with the Session Description Protocol (SDP)".

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

[7] ETSI TS 124 607: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.607 Release 10)".

[8] ETSI TS 124 608: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.608 Release 10)".

[9] ETSI TS 124 604: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication Diversion (CDIV) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.604 version 10.3.0 Release 10)".

[10] ETSI TS 124 605: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Conference (CONF) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.605 Release 10)".

[11] ETSI TS 124 629: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Explicit Communication Transfer (ECT) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.629 Release 10)".

[12] ETSI TS 124 611: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Anonymous Communication Rejection (ACR) and Communication Barring (CB) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.611 Release 10)".

[13] ETSI TS 124 654: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Closed User Group (CUG) using IP Multimedia (IM) Core Network (CN) subsystem, Protocol Specification (3GPP TS 24.654 Release 10)".

[14] ETSI TS 124 642: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Completion of Communications to Busy Subscriber (CCBS) and Completion of Communications by No Reply (CCNR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification (3GPP TS 24.642 Release 10)".

[15] ETSI TS 124 615: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification (3GPP TS 24.615 Release 10)".

[16] ETSI TS 124 606: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Message Waiting Indication (MWI) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.606 Release 10)".

[17] ETSI TS 124 610: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication HOLD (HOLD) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.610 Release 10)".

[18] ETSI TS 124 616: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Malicious Communication Identification (MCID) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.616 Release 10)".

[19] ETSI TS 129 658: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; TISPAN; SIP Transfer of IP Multimedia Service Tariff Information; Protocol specification (3GPP TS 29.658 Release 10)".

[20] ETSI TS 124 628: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.628 Release 10)".

[21] IETF RFC 5009 (September 2007): "Private header (P-Header) extension to the Session Initiation Protocol (SIP) for authorization of Early Media".

[22] ITU-T Recommendation V.152 (November 2004): "Procedures for supporting Voice-Band Data over IP Networks".

[23] ITU-T Recommendation T.38 (September 2010, prepublished): "Procedures for real-time Group 3 facsimile communication over IP networks".

[24] ITU-T Recommendation Q.1912.5: "SERIES Q: SWITCHING AND SIGNALLING Specifications of signalling related to Bearer Independent Call Control (BICC) Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".

[25] ETSI TS 183 036: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ISDN/SIP interworking; Protocol specification".

2.2 Informative references

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] 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]".

[i.2] ISO/IEC 9646 (1994): "Information technology -- Open Systems Interconnection -- Conformance testing methodology and framework".

[i.3] ETSI TR 102 775 (V1.5.1): "Speech and multimedia Transmission Quality (STQ); Guidance on objectives for Quality related Parameters at VoIP Segment-Connection Points; A support to NGN transmission planners".

[i.4] ITU-T Recommendation Q.1902.2 (07/2001): "Bearer Independent Call Control protocol (Capability Set 2) and Signalling System No.7 ISDN User Part: General functions of messages and parameters".

3 Definitions and abbreviations

3.1 Definitions

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

For BICC or ISUP specific terminology, reference shall be made to ITU-T Recommendation Q.1902.2 [i.4]. For SIP and SDP specific terminology, reference shall be made to RFC 3261 [4] and RFC 4566 [3] respectively. Definitions for additional terminology used in this interworking Recommendation are as follows:

Adjacent SIP Node (ASN): SIP node (e.g. SIP Proxy or Back-to-Back User Agent or the SIP side of an IWU) that has established a direct trust relation (association) with Incoming or Outgoing IWU entities

NOTE: The SIP Proxy and Back-to-Back User Agent are defined in accordance with RFC 3261 [4].

Basic Call Control (BCC): signalling protocol associated with the DSS1 - ISDN Basic Call control procedures of ITU-T recommendation Q.931 [15] (EN 300 403-1 [i.1])

Incoming Interworking Unit (I-IWU): physical entity, (which can be combined with a BICC ISN or ISUPexchange) that terminates incoming calls using SIP and originates outgoing calls using the BICC or ISUP protocols

incoming or outgoing: direction of a call (not signalling information) with respect to a reference point

incoming SIP or BICC/ISUP (network): network, from which the incoming calls are received, that uses the SIP or BICC/ISUP protocol (without the term "network", it simply refers to the protocol)

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

Outgoing Interworking Unit (O-IWU): physical entity, (which can be combined with a BICC ISN or ISUP exchange) that terminates incoming calls using BICC or ISUP protocols and originates outgoing calls using the SIP

outgoing SIP or BICC/ISUP (network): network, to which the outgoing calls are sent, that uses the SIP or BICC/ISDN protocol

NOTE: Without the term "network", it simply refers to the protocol.

SIP precondition: indicates the support of the SIP "precondition procedure"

NOTE: as defined in RFC 3312 [6].

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, which 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/IEC 9646 [i.2].

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

3.1.1 Conventions for representation of SIP/SDP information

1) All letters of SIP method names are capitalized.

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 [4]), in plain text without surrounding angle brackets.

EXAMPLE 3: Request-URI, the user info 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.

3.2 Abbreviations

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

ACR	Anonymous Communication Rejection
CB	Communication Barring
CFB	Communication Forwarding Busy
CCBS	Completion of Communications to Busy Subscriber
CCNR	Completion of Communications by No Reply
CD	Communication Deflection
CDIV	Communication DIVersion
CDP	Charging Determinating Point
CDR	Communication Data Record
CFNL	Communication Forwarding Not Logged in
CFNR	Communication Forwarding No Reply
CFU	Communication Forwarding Unconditional
CONF	Conference
CUG	Closed User Group
CW	Communication Waiting
ECT	Explicit Communication Transfer
GW	GateWay
HOLD	Communication Hold
ISDN	Integrated Services Digital Network
IUT	Implementation Under Test
MCID	Malicious Communication Identification
MWI	Message Waiting Indication
OIP	Originating Identification Presentation
OIR	Originating Identification presentation Restriction
PASP	Public Answering Safety Point
PICS	Protocol Implementation Conformance Statement
PSTN	Public Switched Telephone Network
QoS	Quality of service
SIP	Session Initiation Protocol
TIP	Terminating Identification Presentation
TIR	Terminating Identification Restriction
TP	Test Purpose
TSS	Test Suite Structure

4 Test Suite Structure (TSS)

BCALL	successful	SS_bcall_xxx	
	Codec_Negotiation	SS_codec_xxx	
	Resource_Reservation	SS_resource_xxx	
	unsuccessful	SS_unsucc_xxx	
SIP-SIP	Service	OIP	SS_oip_xxx
		OIR	SS_oir_xxx
		TIP	SS_tip_xxx
		TIR	SS_tir_xxx
		HOLD	SS_hold_xxx
		CFU	SS_cfu_xxx
		CFB	SS_cfb_xxx
		CFNR	SS_cfnr_xxx
		CFNL	SS_cfnl_xxx
		CD	SS_cd_xxx
		CONF	SS_conf_xxx
		ACR-CB	SS_acr-cb_xxx
		CUG	SS_cug_xxx
		CW	SS_cw_xxx
		ECT	SS_ect_xxx
		MCID	SS_mcid_xxx
		MWI	SS_mwi_xxx
		CC	SS_cc_xxx
	SIP-I	UUS	SS_uus_xxx
		SUB	SS_sub_xxx
		TP	SS_tp_xxx
	NubP	SS_NP_xxx	
	ACCOUNTING	SS_acc _xxx	
	CS	SS_csel_xxx	
	EmC	SS_ecall_xxx	
	SIP_charging	SS_sipc_xxx	
	SIP-SIP/QoS	SS_qos_xxx	

5 Declarations

5.1 Numbering Scheme

FFS

5.2 Reference configuration

This reference configuration depicted in figure 5.2-1 shall be used to perform an interconnection test between two network operators. Here is depicted the reference point to observe the message flow at the 'Ic' interface between the two networks (in the Testpurposes mentioned '**Interconnection Interface**') one for a single operator and the possible set of end devices used to perform the Test Purposes.

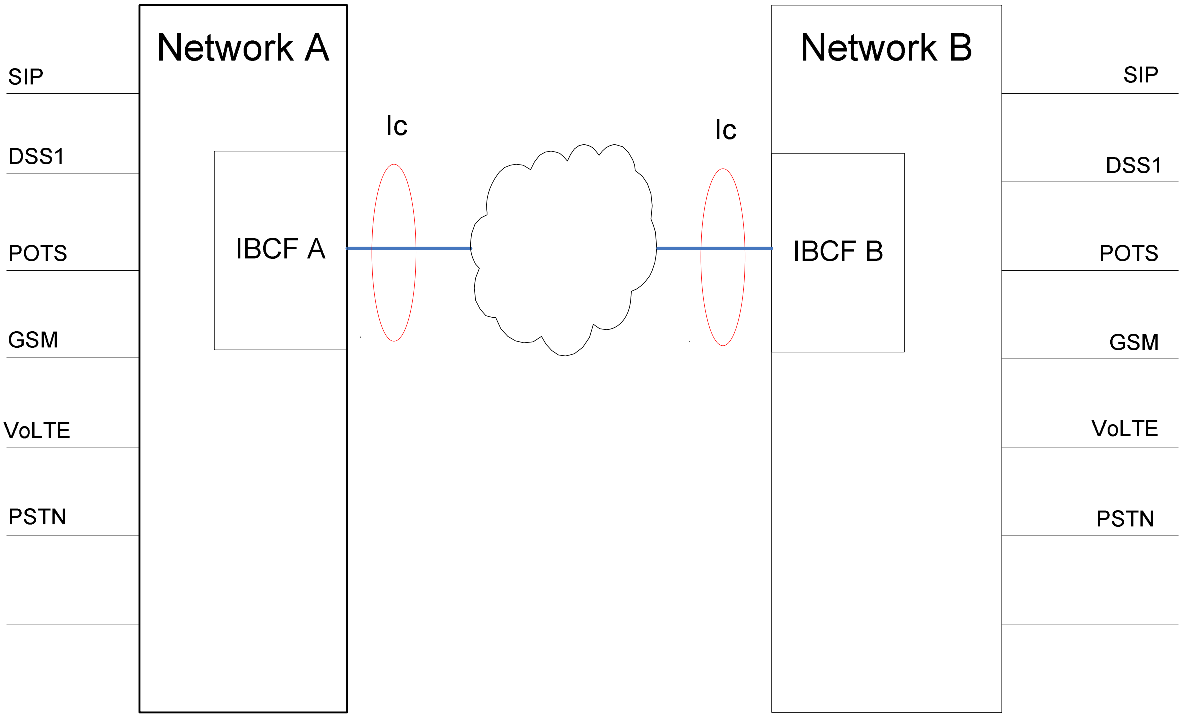

Figure 5.2-1: Reference configuration for the interconnection test

5.3 Selection of End Devices

With the specified Test Purposes in the present document, the compatibility between the interconnected networks and the used end devices in the relevant networks shall be assured. Each Test Purpose shall be performed by using a physical end device to assure the end-to-end compatibility between the two interconnected networks. This is strictly recommended due to the fact that the impact from a end device to another end device is important and will marginal compensated by the network.

Which Test Purposes are possible to perform depends on the types of end devices is used in the relevant network. The table 5.3-1 gives an overview of the end devices used in the relevant networks. The test staff of the network operator decides which type of end device is applicable for the test phase.

The green highlighted element in the table represents the mandatory type of end devices used in the test.

The yellow highlighted elements in the table represents the optional type of end devices used in the test.

Table 5.3-1: Used end devices in the relevant network

Type of End devices	**Network B**						
Network A	**SIP**	**POTS**	**ISDN**	**GSM**	**VoLTE**	**PSTN**	
SIP							
POTS							
ISDN							
GSM							
VoLTE							
PSTN							

6 Selection Expressions

Table 6-1 is used to select the optional Test Purposes for the compatibly test between network operator A and network operator B. The decision whether a Selection Expression id fulfilled is basically agreed regarding the role of the network in the test.

- Network operator 1 is in the role of Network A, Network operator 2 is in the role of Network B.

In case of **Repeat this test in reverse direction.** mentioned in the Comment line in the Test Purpose.

- Network operator 2 is in the role of Network A, Network operator 1 is in the role of Network B.

In each Test Purpose is determined in the field **SELECTION EXPRESSION** whether the selection expression applies and the Test Purpose shall be performed. It has to be decided in which role the Test purpose is applicable (Support Network A, Support Network B).

Before start of test, the table shall be filled out (yes/no) due to the operators gives an answer to the questions in table 6-1. This table can be used as a PICS form as used in a conformance test.

Table 6-1: Selection expression applicable in the Test Purposes

SELECTION EXPRESSION:		Support Network A	Support Network B
Network capabilities			
SE 1:	The originating network (Network A) sends the P-Charging-Vector header		
SE 2:	The originating network (Network A) sends a subset of parameters in the P-Charging-Vector header		
SE 3:	The P-Early-Media header is supported		
SE 4:	Overlap procedure using multiple INVITE method is supported		
SE 5:	Overlap sending using in-dialog method is supported		
SE 6:	Network A supports the PSTN XML schema?		
SE 7:	The resource reservation procedure is supported?		
SE 8:	The Number Portability is supported?		
SE 9:	The network is untrusted?		
SE 10:	Originating network does not have a number portability data base, the number portability look up is done in the interconnected network?		
SE 11:	The network supports the REFER method?		
SE 12:	The Network supports the 3 party call control procedure (REFER interworking)?		
SE 13:	The Number Portability is supported?		
SE 14:	Carrier Selection is performed?		
SE 15:	The Network is a Long distance carrier (Verbindungsnetzbetreiber - VNB)		
SE 16:	SIP Support of Charging is supported?		
SE 17:	The interworking ISUP - SIP I is performed in the network		
Supplementary services			
SE 18:	The network supports the Originating Identification Presentation (OIP)?		
SE 19:	The network supports the "Special arrangement" procedure for the originating user**?**		
SE 20:	The network supports the Originating Identification Restriction (OIR)?		
SE 21:	The Network supports the Terminating Identification Presentation (TIP)?		
SE 22:	The network supports the "Special arrangement" procedure for the terminating user?		
SE 23:	The Network supports the Terminating Identification Restriction (TIR)?		
SE 24:	The Network supports the session HOLD procedure?		
SE 25:	The network supports Communication Forwarding Unconditional (CFU)?		
SE 26:	The network supports Communication Forwarding Busy (CFB)?		
SE 27:	The network supports Communication Forwarding No Reply (CFNR)?		
SE 28:	The Network supports Communication Forwarding Not Logged in (CFNL)		

SELECTION EXPRESSION:		Support	Support
		Network A	Network B
SE 29:	The Network supports Communication Deflection?		
SE 30:	The Network supports the CDIV Notification procedure?		
SE 31:	The Network supports conference (CONF)		
SE 32:	The Network supports the Communication Barring procedure (CB) - (Black list for incoming calls)?		
SE 33:	The Network supports the Anonymous Communication Rejection (ACR)?		
SE 34:	The Network supports the Closed User Group (CUG)?		
SE 35:	The Network supports the Communication Waiting (CW) service?		
SE 36:	The Network supports the $T_{AS\text{-}CW}$ timer?		
SE 37:	The Network supports Explicit Communication Transfer (ECT)?		
SE 38:	The network supports Malicious Communication Identification (MCID)		
SE 39:	The Network supports Message Waiting Indication (MWI)?		
SE 40:	The Network supports Completion of Communications to Busy Subscriber (CCBS)?		
SE 41:	The Network supports Completion of Communications by No Reply (CCNR)		
Terminal capabilities			
SE 42:	The End device (in Network B) establishes an Early dialogue by sending a 183 AND The Network B allows the bearer transmission in the early dialogue		
SE 43:	The End device supports Fax transmission via G.711 codec		
SE 44:	The End device supports Fax transmission via V.152 codec		
SE 45:	The End device supports Fax transmission via m-line T.38 codec		
SE 46:	A SIP end device is used supporting a ISDN user equipment and the PSTN XML Schema is used		
SE 47:	End device is located in the PSTN or PLMN		
SE 48:	The terminating UE supports the from-change tag procedure **and sends a second user identity in an UPDATE request after the dialogue is confirmed**		
SE 49:	The end device performs ECT using the 'Blind/assured transfer'		
SE 50:	The end device performs ECT using the 'Consultative transfer'		
SE 51:	The end device supports the **Resource reservation** procedure		
PSTN/PLMN Supplementary services			
SE 52:	CLIP/CLIR is supported in the PSTN/PLMN part of the network		
SE 53:	COLP/COLR is supported in the PSTN/PLMN part of the network		
SE 54:	HOLD is supported in the PSTN/PLMN part of the network		
SE 55:	CDIV is supported in the PSTN/PLMN part of the network		
SE 56:	CONF/3PTY is supported in the PSTN/PLMN part of the network		
SE 57:	ACR is supported in the PSTN/PLMN part of the network		
SE 58:	CUG is supported in the PSTN/PLMN part of the network		
SE 59:	CW is supported in the PSTN/PLMN part of the network		
SE 60:	ECT is supported in the PSTN/PLMN part of the network		
SE 61:	MCID is supported in the PSTN/PLMN part of the network		
SE 62:	SUB is supported in the PSTN/PLMN part of the network		
SE 63:	UUS is supported in the PSTN/PLMN part of the network		
SE 64:	TP is supported in the PSTN/PLMN part of the network		

7 Test purposes

The application usage procedures in the ATS shall be compliant to TS 129 165 [1], TS 124 229 [2] and RFC 3261 [4]. The validation of the registration procedure is out of scope of the present document.

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

7.1 Testing of SIP protocol requirements

7.1.1 Test purposes for Basic call, Successful

Test case number	SS_bcall_001
Test case group	BCALL/successful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	**Basic call normal call clearing from the called user.** Ensure that call establishment is performed correctly. In the active call state ensure the property of speech. The call is released from the called user.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
	←	**BYE**		
		200 OK BYE	→	

Comments	Establish a communication from network A to Network B **Check:** Ensure the property of speech. **Check:** Are the media streams terminated after the 200 OK BYE was sent? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_002
Test case group	BCALL/successful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	**Basic call normal call clearing from the calling user.** Ensure that call establishment is performed correctly. In the active call state ensure the property of speech. The call is released from the calling user.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
		BYE	→	
	←	200 OK BYE		

Comments	Establish a communication from network A to Network B **Check:** Ensure the property of speech. **Check:** Are the media streams terminated after the 200 OK BYE was sent? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_003
Test case group	BCALL/successful
Reference	8/[1]
SELECTION EXPRESSION	
Test purpose	**Request line in the INVITE.** Ensure that the Request line in the INVITE contains in the userpart the telephone number of the destination user equipment formatted as a 'tel' URI in the global number format and the host portion is set to the host name of the interconnected network. The user URI parameter is present set to 'phone'.
Configuration	
SIP Parameter	INVITE Request line Address of user B @ network B;user=phone
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **INVITE** ➔ Apply post test routine	
Comments	Establish a communication from network A to Network B **Check:** The userpart is in the format of a tel URI in global number format. **Check:** The hostportion is set to the host name of the interconnected network. **Check:** The user parameter is set to phone. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_004
Test case group	BCALL/successful
Reference	5.10/[2]
Testspec Reference	
SELECTION EXPRESSION	SE 1
Test purpose	**P-Charging-Vector header in the INVITE.** Ensure that the P-Charging-Vector header is present in the INVITE establishes a communication between a user of network A and a user of network B and the 'icid-value' and the 'orig-ioi' parameter is present.
Configuration	
SIP Parameter	INVITE P-Charging-Vector: icid-value; orig-ioi
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**	
Comments	Establish a communication from network A to Network B **Check:** The P-Charging-Vector header contains the icid-value parameter. **Check:** The P-Charging-Vector header contains the orig-ioi parameter. Repeat this test in reverse direction.

Test case number	SS_bcall_005
Test case group	BCALL/successful
Reference	5.10/[2]
Testspec Reference	
SELECTION EXPRESSION	SE 2
Test purpose	**P-Charging-Vector header in the INVITE.** Ensure that the P-Charging-Vector header is present in the INVITE establishes a communication between a user of network A and a user of network B and the 'icid-value' or the 'orig-ioi' parameter is present.
Configuration	
SIP Parameter	INVITE P-Charging-Vector: icid-value; orig-ioi
Message flow	**SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** The P-Charging-Vector header contains the icid-value parameter (optional). **Check:** The P-Charging-Vector header contains the orig-ioi parameter (optional). Repeat this test in reverse direction.

Test case number	SS_bcall_006
Test case group	BCALL/successful
Reference	8/[21]
SELECTION EXPRESSION	[Network A] SE 3
Test purpose	**P-Early-Media header support indication in the initial INVITE request.** Ensure that the support of the P-Early.Media header is indicated in the initial INVITE request. A P-Early-Media header is present set to 'supported'.
Configuration	
SIP Parameter	INVITE P-Early-Media: supported SDP
Message flow	**SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** Is a P-Early-Media header is present in the INVITE request? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_bcall_007</td></tr>
<tr><td>Test case group</td><td>BCALL/successful</td></tr>
<tr><td>Reference</td><td>8/[21]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 3 AND [Network B] SE3 AND SE 42</td></tr>
<tr><td>Test purpose</td><td>P-Early-Media header supported early dialogue with 183.

Ensure that an early dialogue is established by sending a 183 Session Progress from Network B and the P-Early-Media header is present authorizes early media.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td>INVITE
P-Early-Media: supported
SDP

183
P-Early-Media: [any value authorizes early media]
SDP</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
INVITE →
← 183 Session Progress
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Establish a communication from network A to Network B
Check: Is a 183 sent from Network B to establish an early dialogue?
Check: A bearer transmission is possible in backward directions.
Repeat this test in reverse direction.</td></tr>
</table>

<table>
<tr><td>Test case number</td><td>SS_bcall_008</td></tr>
<tr><td>Test case group</td><td>BCALL/successful</td></tr>
<tr><td>Reference</td><td>8/[21]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 3 AND [Network B] SE 3</td></tr>
<tr><td>Test purpose</td><td>P-Early-Media header supported early dialogue with 180.

Ensure that an early dialogue is established by sending a 180 Ringing from Network B and the P-Early-Media header is present authorizes early media.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td>INVITE
P-Early-Media: supported
SDP

180
P-Early-Media: [any value authorizes early media]
SDP</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
INVITE →
← 180 Ringing
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Establish a communication from network A to Network B
Check: Is a 183 sent from Network B to establish an early dialogue?
Check: A bearer transmission is possible in backward directions.
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_bcall_009
Test case group	BCALL/successful
Reference	8/[21]
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE 3 AND SE 25 AND SE 30
Test purpose	**P-Early-Media header supported early dialogue with 181.** Ensure that an early dialogue is established by sending a 181 Call Is Being Forwarded from Network B and the P-Early-Media header is present authorizes early media. The Call is forwarded in network B.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes
SIP Parameter	INVITE P-Early-Media: supported SDP 181 P-Early-Media: [any valu authorizes early media]
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Call Is Being Forwarded **Apply post test routine**	
Comments	Establish a communication from network A to Network B **Check:** Is a 181 sent from Network B to establish an early dialogue? Repeat this test in reverse direction.

Test case number	SS_bcall_010
Test case group	BCALL/successful
Reference	8/[21]
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE 3 AND SE 35
Test purpose	**P-Early-Media header supported early dialogue with 182.** Ensure that an early dialogue is established by sending a 182 Queued from Network B and the P-Early-Media header is present authorizes early media. The Call is a waiting call in network B.
Configuration	
SIP Parameter	INVITE P-Early-Media: supported SDP 182 P-Early-Media: [any value authorizes early media]
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Call Is Being Forwarded **Apply post test routine**	
Comments	Establish a communication from network A to Network B **Check:** Is a 181 sent from Network B to establish an early dialogue? Repeat this test in reverse direction.

Test case number	SS_bcall_011
Test case group	BCALL/successful
Reference	5.10/[2]
SELECTION EXPRESSION	
Test purpose	**Record-route header in the INVITE.** Ensure that the Via header is present in the INVITE establishes a communication between a user of network A and a user of network B and the topmost header is set to the IBCF of network A.
Configuration	
SIP Parameter	INVITE Record-Route: <Address of IBCF in network A>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** The topmost Record-Route header or entry contains the address of the IBCF of network A. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_012
Test case group	BCALL/successful
Reference	5.10/[2]
SELECTION EXPRESSION	
Test purpose	**Via header in the INVITE.** Ensure that the Via header is present in the INVITE establishes a communication between a user of network A and a user of network B and the topmost header is set to the IBCF of network A and contains a branch parameter.
Configuration	
SIP Parameter	INVITE Via: <Address of IBCF in network A>; branch=[any value]
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** The topmost Via header contains the Address of IBCF in network A and a branch parameter. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_013
Test case group	BCALL/successful
Reference	5.10/[2]
SELECTION EXPRESSION	
Test purpose	**Record-Route header in the 180 Ringing.** Ensure that the Record-Route header is present in the 180 Ringing provisional response as the first response from network B upon a connection establish setup from network A.
Configuration	
SIP Parameter	180: Record-Route
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** The Record-Route header is present is present in the 180 Ringing. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_014
Test case group	BCALL/successful
Reference	5.10/[2]
SELECTION EXPRESSION	
Test purpose	**Route header in the BYE of the originating user.** Ensure that the Route header is present in the BYE request sent from the originating user equipment in network A and the topmost Route header or entry is set to the IBCF of network B.
Configuration	
SIP Parameter	BYE: Route: <Address of IBCF in network B>;lr, ….
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** **A confirmed session already exists** BYE → ← 200 OK BYE **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** The Route header is present is present in the BYE and the topmost header or entry is set to the address of the IBCF of network B. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_015
Test case group	BCALL/successful
Reference	5.10/[2]
SELECTION EXPRESSION	
Test purpose	**Route header in the BYE of the terminating user.** Ensure that the Route header is present in the BYE request sent from the terminating user equipment in network B and the topmost Route header or entry is set to the IBCF of network A.
Configuration	
SIP Parameter	BYE: Route: <Address of IBCF in network A>;lr, ….
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **A confirmed session already exists** ← BYE 200 OK BYE → **Apply post test routine**	
Comments	Establish a communication from network A to Network B **Check:** The Route header is present is present in the BYE and the topmost header or entry is set to the address of the IBCF of network A. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_016
Test case group	BCALL/successful
Reference	5.10/[2]
SELECTION EXPRESSION	
Test purpose	**Route header in the ACK.** Ensure that the Route header is present in ACK from network A upon a connection establishment from network A is completed and the topmost Route header or entry is set to the IBCF of network B.
Configuration	
SIP Parameter	ACK: Route: <Address of IBCF in network B>;lr, ….
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing ← 200 OK INVITE ACK → **Apply post test routine**	
Comments	Establish a communication from network A to Network B **Check:** Route header is present is present in the ACK and the topmost header or entry is set to the address of the IBCF of network B. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_017
Test case group	BCALL/successful
Reference	[4] and [5]
SELECTION EXPRESSION	
Test purpose	**Handling of SDP parameters in the INVITE.** Ensure that call establishment and the correct handling 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 7.1.1-1 applies.
Configuration	
SIP Parameter	INVITE: Content-Type: application/sdp m=audio <Port number> RTP/AVP TYPE_SDP= PIXIT (table 7.1.1-1) *or* m= Image <Port number> Udptl *or* Tcptl TYPE_SDP= PIXIT (table 7.1.1-1) a=TYPE_SDP= PIXIT (table 1) b=TYPE_SDP= PIXIT (table 1)
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** Is the preferred codec set to TYPE_SDP? **Check:** If present: is the a line set to TYPE_SDP? **Check:** If present: is the b line set to TYPE_SDP? **Check:** Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_018
Test case group	BCALL/successful
Reference	[4] and [5]
SELECTION EXPRESSION	
Test purpose	**The SDP answer is sent in the 200 OK.** Ensure that the call establishment performed correctly. The initial INVITE contains a SDP with the offer 1 according table 7.1.1-1. Ensure that answer related to the SDP offer is contained in the 200 OK INVITE message. Ensure that in the confirmed state the voice transfer on the media and B-channels is performed correctly.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)** Apply post test routine	**Interconnection Interface** **SIP (Network B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK →
Comments	Establish a communication from network A to Network B **Check:** Is the SDP answer contained in the 200 OK INVITE. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_018
Test case group	BCALL/successful
Reference	[4] and [5]
SELECTION EXPRESSION	
Test purpose	**First response 200 OK INVITE.** Ensure that call establishment and the correctly if the called user answers with a 200 OK message.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 200 OK INVITE ACK → **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** Is it possible to confirm a session without early dialogue? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Table 7.1.1-1

TYPE_SDP	**m= line**			**b= line**	**a= line**
VA	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth-value> (see note)**	**rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>**
VA_01	Audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A or rtpmap 0 PCMU/8000
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 or rtpmap 8 PCMA/8000
VA_04	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000
VA_05	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> CLEARMODE
NOTE: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.					

Test case number	SS_bcall_020
Test case group	BCALL/successful
Reference	[4] and [5]
SELECTION EXPRESSION	[Network A] SE 43 AND [Network B] SE 43
Test purpose	**Fax transmission using the G.711 codec.** Ensure that a Fax transmission is possible from Network A to Network B and the relevant codec is the G.711 codec. Ensure in the active call state the property of Fax transmission.
Configuration	
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** Is the SDP answer contained in the 200 OK INVITE. **Check:** is Fax transmission is successful? Repeat this test in reverse direction.

Test case number	SS_bcall_021
Test case group	BCALL/successful
Reference	[5] and [22]
SELECTION EXPRESSION	[Network A] SE 44 AND [Network A] SE 44
Test purpose	**Fax transmission using the V.152 codec.** Ensure that a Fax transmission is possible from Network A to Network B and the relevant codec is the V.152 codec. Ensure in the active call state the property of Fax transmission.
Configuration	
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8 <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** Contains the SDP offer in the initial INVITE a voice band data codec. **Check:** contains the SDP answer in the 180 or 200 OK INVITE a voice band data codec. **Check:** Is Fax transmission is successful? Repeat this test in reverse direction.

Test case number	SS_bcall_022
Test case group	BCALL/successful
Reference	[5] and [23]
SELECTION EXPRESSION	[Network A] SE 45 AND [Network B] SE 45
Test purpose	**Fax transmission using the T.38 in an audio m-line codec.** Ensure that a Fax transmission is possible from Network A to Network B and the relevant codec is the T.38 in an 'audio' m-line codec. Ensure in the active call state the property of Fax transmission.
Configuration	
SIP Parameter	INVITE: SDP m=image <Port> udptl t38 180/200 OK INVITE: SDP m=image <Port> udptl t38
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** Contains the SDP offer in the initial INVITE a T.38 codec in an 'audio' line. **Check:** Contains the SDP answer in the 180 or 200 OK INVITE a T.38 codec in an 'audio' line. **Check:** Is Fax transmission is successful? Repeat this test in reverse direction.

Test case number	SS_bcall_023
Test case group	BCALL/successful
Reference	4.9, N/[2]
SELECTION EXPRESSION	[Network A] SE 47 AND [Network A] SE 4 AND [Network B] SE 4
Test purpose	**Overlap sending, the Multiple INVITE method is used.** Ensure that call establishment using overlap sending is performed correctly. Ensure that in the confirmed state the voice transfer on the media and B-channels is performed correctly.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE(CSq 1)	→	
		INVITE(CSq 2)	→	
	←	484 Address Incomplete(CSq 1)		
		ACK	→	
		INVITE(CSq 3)	→	
	←	484 Address Incomplete(CSq 2)		
		ACK	→	
		…….		
		INVITE(CSq 4)	→	
	←	484 Address Incomplete(CSq 3)		
		ACK	→	
	←	180 Ringing(CSq 4)		
		Apply post test routine		

Comments	Establish a communication from ISDN to SIP using the overlap operation in ISDN **Check:** All INVITE requests contains the same Call ID and From header values. SIP answers with 180 Ringing. Repeat this test in reverse direction.

Test case number	SS_bcall_024
Test case group	BCALL/successful
Reference	4.9, N/[2]
SELECTION EXPRESSION	[Network A] SE 47 AND [Network A] SE 4 AND [Network B] SE 5
Test purpose	**Overlap sending, the in-Dialogue method is used** Ensure that call establishment using overlap sending is performed correctly. Ensure that in the confirmed state the voice transfer on the media and B-channels is performed correctly.
Configuration	
SIP Parameter	INVITE 2: Supported: 100rel 183: Require: 100rel INFO: Content-Type: application/x-session-info SubsequentDigit: <additional digits>
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(CSq 1) 1	→	
	←	484 Address Incomplete(CSq 1)		
		ACK	→	
		INVITE(CSq 2) 2	→	
	←	183 Session Progress(CSq 2)		
		PRACK	→	
	←	200 OK PRACK		
		INFO	→	
	←	200 OK INFO		
		……..		
		INFO	→	
	←	200 OK INFO		
	←	180 Ringing(CSq 2)		
		Apply post test routine		

Comments	Establish a communication from ISDN to SIP using the overlap operation in ISDN **Check:** All INVITE requests contains the same Call ID and From header values. **Check:** The 183 session Progress that establishes an early dialogue contains a Require header set to 100rel. **Check:** All INFO requests contain the Content-Type header set to 'application/x-session-info'. **Check:** All INFO requests contains the 'SubsequentDigit:' MIME body containing the additional digits. The UE B answers with 180 Ringing response after the INVITE was received. Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_bcall_025</td></tr>
<tr><td>Test case group</td><td>BCALL/successful</td></tr>
<tr><td>Reference</td><td>5.1.1.1.2/[25]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] (SE 46 OR SE 47) AND [Network A] SE 6</td></tr>
<tr><td>Test purpose</td><td>PSTN XML BearerCapability element in the INVITE.

User A is located in network A and an ISDN end device is used. Ensure that the INVITE request contains a PSTN XML MIME body and a BearerCapability element as indicated in table 7.1.1-2 is present.</td></tr>
<tr><td>Configuration</td><td>User A is an ISDN access either in the PSTN or the SIP - ISDN interworking according [10] applies</td></tr>
<tr><td>SIP Parameter</td><td>INVITE:
Content-Type: application/vnd.etsi.pstn+xml
Content-Disposition: signal;handling=optional

<?xml version="1.0" encoding="utf-8"?>
PSTN
BearerCapability
BCoctet3
CodingStandard>00<
InformationTransferCabability>ITC_value<
< BCoctet4
TransferMode>00<
InformationTransferRate>10000<
BCoctet5
Layer1Identification>01<
UserInfoLayer1Protocol>00011<</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
INVITE ➔
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Check: Is a PSTN XML MIME body contained in the INVITE request?
Check: Is the BearerCapability element is present?
Check: Is InformationTransferCabability element is set as indicated in table 2.1.1-1?
Check: Is the InformationTransferCabability element value consistent with the codec list in the SDP?
Check: Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP?
Repeat this test in reverse direction.</td></tr>
</table>

Table 7.1.1-2: PSTN XML BearerCapability

ITC_value	BC Information transfer capability	XML InformationTransferCabability
ITC_VA_1	Speech	'00000'
ITC_VA_2	3,1 kHz audio	'10000'
ITC_VA_3	unrestricted digital information	'01000'

Test case number	SS_bcall_026
Test case group	BCALL/successful
Reference	5.1.1.1.2/[25]
SELECTION EXPRESSION	[Network A] (SE 46 OR SE 47) AND [Network A] SE 6
Test purpose	**PSTN XML HighLayerCapability element in the INVITE.** User A is located in network A and an ISDN end device is used. Ensure that the INVITE request contains a PSTN XML MIME body and a HighLayerCapability element is present.
Configuration	User A is an ISDN access either in the PSTN or the SIP - ISDN interworking according [10] applies
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN HighLayerCompatibility HLOctet3 CodingStandard>00< Interpretation>100< PresentationMethod>01< HLOctet4 HighLayerCharacteristics>[any value]<
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE → **Apply post test routine**	
Comments	**Check:** Is a PSTN XML MIME body contained in the INVITE request? **Check:** Is the HighLayerCapability element is present? Repeat this test in reverse direction.

Test case number	SS_bcall_027
Test case group	BCALL/successful
Reference	5.1.1.1.2/[25]
SELECTION EXPRESSION	[Network A] (SE 46 OR SE 47) AND [Network A] SE 6
Test purpose	**PSTN XML ProgressIndicator element in the INVITE.** User A is located in network A and an ISDN end device is used. Ensure that the INVITE request contains a PSTN XML MIME body and at least one ProgressIndicator element is present.
Configuration	User A is an ISDN access either in the PSTN or the SIP - ISDN interworking according [10] applies
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN ProgressIndicator ProgressOctet3 CodingStandard>00< Location>yyyy< ProgressOctet4 ProgressDescription>0000110< *ProgressIndicator* *ProgressOctet3* *CodingStandard>00<* *Location>0000<* *ProgressOctet4* *ProgressDescription>[any value]<*
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**	
Comments	**Check:** Is a PSTN XML MIME body contained in the INVITE request? **Check:** Is a ProgressIndicator element present and the ProgressDescription element is set to '0000110'? **Check:** Is optional a second ProgressIndicator element present and the ProgressDescription element is set to any value not #2 and not #8? Repeat this test in reverse direction.

Test case number	SS_bcall_028
Test case group	BCALL/successful
Reference	5.1.2.2/[25]
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6
Test purpose	**PSTN XML ProgressIndicator element in the 180.** User B is located in network B and an ISDN end device is used. Ensure that the 180 Ringing response contains a PSTN XML MIME body and at least one ProgressIndicator element is present.
Configuration	User B is an ISDN access either in the PSTN or the SIP - ISDN interworking according [10] applies
SIP Parameter	180: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN ProgressIndicator ProgressOctet3 CodingStandard>00< Location>yyyy< ProgressOctet4 ProgressDescription>0000111< *ProgressIndicator* *ProgressOctet3* *CodingStandard>00<* *Location>0000<* *ProgressOctet4* *ProgressDescription>[any value]<*
Message flow	**SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing **Apply post test routine**
Comments	**Check:** Is a PSTN XML MIME body contained in the 180 Ringing response? **Check:** Is a ProgressIndicator element present and the ProgressDescription element is set to '0000110'? **Check:** Is optional a second ProgressIndicator element present and the ProgressDescription element is set to any value not #2 and not #8? Repeat this test in reverse direction.

Test case number	SS_bcall_029
Test case group	BCALL/successful
Reference	5.1.2.3/[25]
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6
Test purpose	**PSTN XML ProgressIndicator element in the 200.** User B is located in network B and an ISDN end device is used. Ensure that the 200 OK INVITE response contains a PSTN XML MIME body and at least one ProgressIndicator element is present.
Configuration	User B is an ISDN access either in the PSTN or the SIP - ISDN interworking according [10] applies

SIP Parameter	200:
	    Content-Type: application/vnd.etsi.pstn+xml
	    Content-Disposition: signal;handling=optional
	
	<?xml version="1.0" encoding="utf-8"?>
	PSTN
	    ProgressIndicator
	        ProgressOctet3
	            CodingStandard>00<
	            Location>yyyy<
	        ProgressOctet4
	            ProgressDescription>0000111<

Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing ← 200 OK INVITE ACK → **Apply post test routine**	
Comments	**Check:** Is a PSTN XML MIME body contained in the 200 OK INVITE response? **Check:** Is a ProgressIndicator element present and the ProgressDescription element is set to '0000110'? Repeat this test in reverse direction.

Test case number	SS_bcall_030
Test case group	BCALL/successful
Reference	5.1.1.1.2/[25]
SELECTION EXPRESSION	[Network A] (SE 46 OR SE 47) AND [Network A] SE 6
Test purpose	**PSTN XML BearerCapability Fallback connection type element in the INVITE.** User A is located in network A and an ISDN end device is used. Ensure that the INVITE request contains a PSTN XML MIME body and one BearerCapability element is present the InformationTransferCabability element is set to '00000' and one InformationTransferCabability element is set to '10001'.
Configuration	User A is an ISDN access either in the PSTN or the SIP - ISDN interworking according [10] applies
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>00000< BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>10001<
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**	
Comments	**Check:** Is a PSTN XML MIME body contained in the INVITE request? **Check:** Is the first BearerCapability InformationTransferCabability element is set as indicated to '00000'? **Check:** Is the second BearerCapability InformationTransferCabability element is set as indicated to '10001'? **Check:** Is the InformationTransferCabability element value consistent with the codec list in the SDP? **Check:** Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP? Repeat this test in reverse direction.

Test case number	SS_bcall_031
Test case group	BCALL/successful
Reference	5.1.2.3/[25]
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6
Test purpose	**Fall back does not occur.** User B is located in network B and an ISDN end device is used. The Fallback connection type was requested in the initial INVITE request. Ensure that the 200 OK INVITE response contains a PSTN XML MIME body and a BearerCapability element is present the InformationTransferCabability element set to '10001'.
Configuration	User B is an ISDN access either in the PSTN or the SIP - ISDN interworking according [10] applies
SIP Parameter	200: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>10001<
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing ← 200 OK INVITE ACK → **Apply post test routine**
Comments	**Check:** Is a PSTN XML MIME body contained in the 200 OK INVITE response? **Check:** Is a BearerCapability element present, the InformationTransferCabability element set to '10001'? **Check:** Is the InformationTransferCabability element value consistent with the codec list in the SDP? **Check:** Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP? Repeat this test in reverse direction.

Test case number	SS_bcall_032
Test case group	BCALL/successful
Reference	5.1.2.3/[25]
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6
Test purpose	**Fall back occurs.** User B is located in network B and an ISDN end device is used. The Fallback connection type was requested in the initial INVITE request. Ensure that the 200 OK INVITE response contains a PSTN XML MIME body and a BearerCapability element is present the InformationTransferCabability element set to '00000'. A PSTN XML MIME ProgressIndicator body is present, the ProgressDescription is set to '0000101'.
Configuration	User B is an ISDN access either in the PSTN or the SIP - ISDN interworking according [10] applies
SIP Parameter	200: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>00000< ProgressIndicator ProgressOctet4 ProgressDescription>0000101<
Message flow	**SIP (Network A)** — **Interconnection Interface** — **SIP (Network B)** INVITE → ← 180 Ringing ← 200 OK INVITE ACK → **Apply post test routine**
Comments	**Check:** Is a PSTN XML MIME body contained in the 200 OK INVITE response? **Check:** Is a BearerCapability element present, the InformationTransferCabability element set to '00000'? **Check:** Is a ProgressIndicator element is present, the ProgressDescription is set to '0000101'? **Check:** Is the InformationTransferCabability element value consistent with the codec list in the SDP? **Check:** Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP? Repeat this test in reverse direction.

Test case number	SS_bcall_033
Test case group	BCALL/successful
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47
Test purpose	**SIP-I support, Basic call, IAM present in the INVITE request.** Ensure that when a call initiated in the PSTN or the PLMN and the ISUP - SIP-I interworking is applicable in the originating network, a ISUP IAM is encapsulated in the initial INVITE request. Ensure that all the mandatory parameters in the IAM are present and the values are valid and the **Transmission medium requirement** parameter is consistent with the SDP.
Configuration	
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** **Nature of connection indicators** **Forward call indicators** **Calling party's category** **Transmission medium requirement** **Called party number** *Calling party number (optional)* *Optional forward call indicators (optional)* *Hop counter (optional)* *User service information (optional)* *Access transport (optional)* --[any boundary name]--
Message flow	
SIP (Network A) **Interconnection Interface** **SIP (Network B)** INVITE(IAM) → ← 100 Trying **Apply post test routine**	
Comments	Establish a communication from network A to Network B **Check:** Is an ISUP IAM encapsulated in the INVITE request? **Check:** Are all the mandatory ISUP parameters present in the IAM and are the values valid? **Check:** Are the values of the optional parameters in the encapsulated IAM valid? **Check:** Is the 'm' line with corresponding attributes in the SDP consistent with the **Transmission medium requirement** parameter? **Check:** Is the **Transmission medium requirement** value consistent with the bandwidth information in the SDP? Repeat this test in reverse direction.

Test case number	SS_bcall_034
Test case group	BCALL/successful
Reference	7.2.1/[24]
SELECTION EXPRESSION	[Network A] SE 4 AND SE 17 AND SE 47
Test purpose	**SIP-I support, Basic call, overlap signalling.** Ensure that when overlap signalling applies in the ISUP -SIP-I interworking in the originating network, several INVITE requests with the same Cal-ID and From tag are sent from Network A to Network B. Ensure that the original IAM is encapsulated in any INVITE request.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE(1)	→	
	←	484 Address Incomplete(1)		
		ACK	→	
		INVITE(2)	→	
	←	484 Address Incomplete(2)		
		ACK	→	
		INVITE(3)	→	
	←	484 Address Incomplete(3)		
		ACK	→	
		.		
		.		
		INVITE(4)	→	
	←	180 Ringing(4)		
		Apply post test routine		

Comments	Establish a communication from network A to Network B using the overlap procedure in Network A **Check:** Are the INVITE requests sent with the same From tag and the Call-ID? **Check:** After the 180 applies, are all previous INVITE transactions are terminated with a 484 final response? **Check:** Is the encapsulated IAM present in the initial INVITE request also encapsulated in any following INVITE request required for the call setup? Repeat this test in reverse direction.

Test case number	SS_bcall_035
Test case group	BCALL/successful
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	**SIP-I support, Basic call, ACM present in the 180 response.** Ensure that on receipt of a 180 Ringing provisional response and an SIP-I - ISUP interworking is applicable in the terminating network the Backward call indicators parameter in the encapsulated ACM is present and the values are valid. Ensure that the values of the optional parameters in the encapsulated ACM are valid.
Configuration	
SIP Parameter	180: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** **Backward call indicators** --[any boundary name]--
Message flow	
SIP (Network A) / **Interconnection Interface** / **SIP (Network B)**	INVITE → ← 100 Trying ← 180 Ringing(ACM) **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** Is an ISUP ACM message encapsulated in the 180 Ringing provisional response? **Check:** Is the mandatory Backward call indicators parameter present in the encapsulated ISUP ACM and are the values valid? **Check:** Are the values of optional parameters in the encapsulated ISUP ACM valid? **Check:** If an SDP answer is present in the 180, are the codec and the bandwidth information in the 'a' attributes consistent with Transmission medium requirement in the encapsulated IAM of the INVITE request? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td colspan="2">SS_bcall_036</td></tr>
<tr><td>Test case group</td><td colspan="2">BCALL/successful</td></tr>
<tr><td>Reference</td><td colspan="2">6.5/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td colspan="2">[Network B] SE 17 AND SE 47</td></tr>
<tr><td>Test purpose</td><td colspan="2">SIP-I support. Basic call, early ACM present in the 183 response.

Ensure that on receipt of a 183 Session Progress provisional response and an SIP-I - ISUP interworking is applicable in the terminating network the Backward call indicators parameter in the encapsulated ACM is present and the value of the Called party's status indicator is set to 'no indication'.
Ensure that the values of the optional parameters in the encapsulated ACM are valid.</td></tr>
<tr><td>Configuration</td><td colspan="2">Select a proper destination that sends an early ACM in the PSTN/PLMN e.g. announcement</td></tr>
<tr><td>SIP Parameter</td><td colspan="2">183:
Content-Type: multipart/mixed;boundary=[any boundary name]

--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

ACM
Backward call indicators
Called party's status indicator= no indication

--[any boundary name]--</td></tr>
<tr><td colspan="3">Message flow</td></tr>
<tr><td>SIP (Network A)</td><td>Interconnection Interface
INVITE →
← 100 Trying
← 183 Session Progress(ACM)
Apply post test routine</td><td>SIP (Network B)</td></tr>
<tr><td>Comments</td><td colspan="2">Establish a communication from network A to Network B
Check: Is an ISUP ACM message encapsulated in the 183 Session Progress provisional response?
Check: Is the mandatory Backward call indicators parameter present in the encapsulated ISUP ACM and are the values valid?
Check: Is the Called party's status indicator in the encapsulated ISUP ACM set to 'no indication'?
Check: Are the values of optional parameters in the encapsulated ISUP ACM valid?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_bcall_037
Test case group	BCALL/successful
Reference	6.6/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	**SIP-I support. Basic call, CPG present in a 180 response.** Ensure that on receipt of a 180 Ringing provisional response and an SIP-I - ISUP interworking is applicable in the terminating network the Event indicator in the encapsulated CPG is present and set to 'ALERTING'. Ensure that the values of the optional parameters in the encapsulated CPG are valid.
Configuration	Select a proper destination that sends at first an early ACM and after then a CPG 'ALERTING' in the PSTN/PLMN (e.g. PBX).
SIP Parameter	180: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** **Event indicator** = ALERTING --[any boundary name]--
Message flow	
SIP (Network A)	**Interconnection Interface** **SIP (Network B)**
	INVITE →
	← 100 Trying
	← 183 Session Progress(ACM)
	← 180 Ringing(CPG)
	Apply post test routine
Comments	Establish a communication from network A to Network B **Check:** Is an ISUP CPG message encapsulated in the 180 Ringing provisional response? **Check:** Is the mandatory Event indicator present in the encapsulated ISUP CPG set to 'ALERTING'? **Check:** Are the values of optional parameters in the encapsulated ISUP CPG valid? Repeat this test in reverse direction.

Test case number	SS_bcall_038
Test case group	BCALL/successful
Reference	6.7/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	**SIP-I support. Basic call, ANM present in a 200 OK INVITE response.** Ensure that on receipt of a 200 OK INVITE final response and an SIP-I - ISUP interworking is applicable in the terminating network the ISUP ANM is encapsulated in the 200 OK. Ensure that the values of the optional parameters in the encapsulated ANM are valid.
Configuration	
SIP Parameter	180: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** --[any boundary name]--
Message flow	
SIP (Network A) / **Interconnection Interface** / **SIP (Network B)**	INVITE → ← 100 Trying ← 180 Ringing(ACM) ← 200 OK INVITE(ANM) ACK → **Apply post test routine**
Comments	Establish a confirmed communication from network A to Network B **Check:** Is an ISUP ANM encapsulated in the 200 OK INVITE? **Check:** Are the values of optional parameters in the encapsulated ISUP ANM valid? **Check:** Ensure the property of speech. **Check:** Are the codec and the bandwidth information in the 'a' attributes consistent with Transmission medium requirement in the encapsulated IAM of the INVITE request? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td colspan="2">SS_bcall_039</td></tr>
<tr><td>Test case group</td><td colspan="2">BCALL/successful</td></tr>
<tr><td>Reference</td><td colspan="2">5.4.3.4, 6.11.2/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td colspan="2">[Network A] SE 17 AND SE 47</td></tr>
<tr><td>Test purpose</td><td colspan="2">SIP-I support. Basic call, REL present in a BYE request sent from the originating network.

Ensure that a ISUP REL message is encapsulated in a BYE request sent in the release procedure initiated from the originating user when ISUP - SIP-I interworking is applicable in the originating network.
Ensure the validity of the cause indicator in the encapsulated REL.
Ensure that the ISUP RLC is encapsulated in the 200 OK BYE.</td></tr>
<tr><td>Configuration</td><td colspan="2"></td></tr>
<tr><td>SIP Parameter</td><td colspan="2">BYE:
Content-Type: multipart/mixed;boundary=[any boundary name]

--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

REL
Cause value:

--[any boundary name]--

200 OK BYE
Content-Type: multipart/mixed;boundary=[any boundary name]

--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

RLC

--[any boundary name]--</td></tr>
<tr><td colspan="3">Message flow</td></tr>
<tr><td>SIP (Network A)</td><td>Interconnection Interface
INVITE →
← 100 Trying
← 180 Ringing
← 200 OK INVITE
ACK →
Communication
BYE(REL) →
← 200 OK BYE(RLC)</td><td>SIP (Network B)</td></tr>
<tr><td>Comments</td><td colspan="2">Establish a confirmed communication from network A to Network B
The originating user terminates the communication
Check: Is the ISUP REL encapsulated in the BYE request?
Check: Are the cause indicators in the encapsulated ISUP REL valid?
Check: If a Reason header is present in the BYE request, is the 'cause' value of Reason header equal to the 'Cause value' in the encapsulated REL?
Check: Is the ISUP RLC encapsulated in the 200 OK BYE?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_bcall_040
Test case group	BCALL/successful
Reference	5.4.3.4, 6.11.2/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	**SIP-I support. Basic call, REL present in a BYE request sent from the terminating network.** Ensure that a ISUP REL message is encapsulated in a BYE request sent in the release procedure initiated from the terminating user when SIP-I - ISUP interworking is applicable in the terminating network. Ensure the validity of the cause indicator in the encapsulated REL. Ensure that the ISUP RLC is encapsulated in the 200 OK BYE.
Configuration	
SIP Parameter	BYE: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **REL** Cause value: --[any boundary name]-- 200 OK BYE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **RLC** --[any boundary name]--
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE	→	
	←	100 Trying		
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
	←	BYE(REL)		
		200 OK BYE(RLC)	→	

Comments	Establish a confirmed communication from network A to Network B The terminating user terminates the communication **Check:** Is the ISUP REL encapsulated in the BYE request? **Check:** Are the cause indicators in the encapsulated ISUP REL valid? **Check:** If a Reason header is present in the BYE request, is the 'cause' value of Reason header equal to the 'Cause value' in the encapsulated REL? **Check:** Is the ISUP RLC encapsulated in the 200 OK BYE? Repeat this test in reverse direction.

7.1.2 Codec negotiation

Test case number	SS_codec_001
Test case group	BCALL/Codec_Negotiation
Reference	[3], [4] and [5]
SELECTION EXPRESSION	
Test purpose	**Session update requested by the calling user.** During the session, the calling user decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE or UPDATE containing a new media description. This re-INVITE or UPDATE 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 7.1.2-1 applies.
Configuration	
SIP Parameter	SDP1: codec x chosen from table 7.1.2-1 SDP3: codec y chosen from table 7.1.2-1
Message flow **SIP (Network A)** **CASE A** **CASE B**	 **Interconnection Interface** **SIP (Network B)** **A confirmed session already exists (SDP 1)** INVITE(SDP3) → ← 200 OK INVITE(SDP4) ACK → UPDATE(SDP3) → ← 200 OK UPDATE(SDP4) **Apply post test routine**
Comments	Establish a communication from network A to Network B using SDP1 chosen from the table 7.1.2-1 **Check:** The calling user changes the media description using INVITE request containing SDP 3 codec chosen from table 7.1.2-1 different to SDP1. **Check:** Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description? Repeat this test in reverse direction.

Test case number	SS_codec_002
Test case group	BCALL/Codec_Negotiation
Reference	[3], [4] and [5]
SELECTION EXPRESSION	
Test purpose	**Session update requested by the called user.** 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 7.1.2-1 applies.
Configuration	
SIP Parameter	SDP1: codec x chosen from table 7.1.2-1 SDP2: codec y chosen from table 7.1.2-1
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** **A confirmed session already exists (SDP 1)** **CASE A** INVITE(SDP3) → ← 200 OK INVITE(SDP4) ACK → **CASE B** UPDATE(SDP3) → ← 200 OK UPDATE(SDP4) **Apply post test routine**
Comments	Establish a connection from SIP UE 1 to SIP UE 2 using SDP1 chosen from the table 7.1.2-1 **Check:** The called user changes the media description using INVITE request containing SDP 2 codec chosen from table 7.1.2-1 different to SDP1. **Check:** Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description? Repeat this test in reverse direction.

Test case number	SS_codec_003
Test case group	BCALL/Codec_Negotiation
Reference	[3], [4] and [5]
SELECTION EXPRESSION	
Test purpose	**The SDP answer is contained in a 200 OK final response.** 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 confirmed call state the voice transfer on the media channels is performed correctly.
Configuration	
SIP Parameter	INVITE: SDP offer 200: SDP answer
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(SDP1) → ← 180 Ringing ← 200 OK INVITE(SDP2) ACK → **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** Is the SDP offer contained in the initial INVITE request? **Check:** Is the SDP answer contained in the 200 OK INVITE final response? Repeat this test in reverse direction.

Table: 7.1.2-1

VARIABLE	PT	Encoding	media type	clock rate	channels	Supported in network A	Supported in network B
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	1 8,000	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			
VA_33	Dyn	AMR	A	8,000	1		
VA_34	Dyn	AMR-WB	A	16,000	1		
VA_35	Dyn	telephone-event	A	8000	1		

7.1.3 Resource Reservation

Test case number	SS_resource_001
Test case group	BCALL/Resource_Reservation
Reference	[3], [4], [5] and [6]
SELECTION EXPRESSION	([Network A] SE 50 AND [Network B] SE 50) AND SE 7
Test purpose	**Resource reservation successful, segmented status.** Ensure that the network is able to reserve resources for quality of service when requested from the initiating user. • In the INVIT the UE requests to establish QoS preconditions for all the media streams. • In the 183 Session Progress the UAS supports the QoS preconditions and requests that UAC sends a confirmation when the QoS preconditions are met. • The UPDATE includes in the SDP the information about the successful QoS bidirectional mode, due to the successful bidirectional PDP context established. • 200 OK UPDATE the SDP contains an indication that the UE successfully reserved the QoS in the send and receive directions.
Configuration	
SIP Parameter	INVITE: Supported: 100rel precondition SDP1: m=audio 3456 RTP/AVP 8 a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv 183 Session Progress: Supported: 100rel precondition SDP2: m=audio 6544 RTP/AVP 8 a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv UPDATE SDP3: m=audio 3456 RTP/AVP 8 a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv 200 OK UPDATE SDP4: a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(SDP1) → ← 183 Session Progress(SDP2) PRACK → ← 200 OK PRACK **Resource reservation** UPDATE(SDP3) → ← 200 OK UPDATE(SDP4) **Apply post test routine**
Comments	Establish a communication from network A to Network B **Check:** Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the **INVITE**? **Check:** Is the quality of service for the desired state local and remote set to 'mandatory' and 'sendrecv' in the **183**? **Check:** Is the quality of service for the current state local set to 'sendrecv' indicated in the SDP in the **UPDATE**? **Check:** Is the quality of service for the current state local and remote set to 'sendrecv' indicated in the SDP in the **200 OK UPDATE**? Repeat this test in reverse direction.

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

Test case number	SS_unsucc_001
Test case group	BCALL/unsuccessful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	**Called number is not allocated in the assumed network.** Ensure that, when calling to unallocated number, the network initiate call clearing to the calling user with a 404 Not Found message.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 404 Not Found ACK →
Comments	Establish a communication from network A to Network B, called user number is not allocated in Network B **Check:** Is a 404 Not Found sent from Network B to Network A? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_unsucc_002
Test case group	BCALL/unsuccessful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	**The network B is unable to process the request.** 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.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 503 Service unavailable ACK →
Comments	Establish a communication from network A to Network B, Network B is unable to process the request. **Check:** Is a 503 Service unavailable sent from Network B to Network A? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_unsucc_003
Test case group	BCALL/unsuccessful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	**The called user is network determined busy.** Ensure that, when the called user is busy, the network initiates call clearing to the calling user with a 486 Busy Here message.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 486 Busy Here ACK →
Comments	Establish a communication from network A to Network B, user B is network determined user busy. **Check:** Is a 486 Busy Here sent from Network B to Network A? Repeat this test in reverse direction.

Test case number	SS_unsucc_004
Test case group	BCALL/unsuccessful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	**The called user is user determined busy.** Ensure that, when the called user is busy, the user initiates call clearing to the calling user with a 486 Busy Here message.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 486 Busy Here ACK →
Comments	Establish a communication from network A to Network B, user B is user determined user busy. **Check:** Is a 486 Busy Here sent from Network B to Network A Repeat this test in reverse direction.

Test case number	SS_unsucc_005
Test case group	BCALL/unsuccessful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	**The called user is not available under the called number.** Ensure that when the number is changed, the network initiate call clearing to the calling user with a 410 Gone message.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 410 Gone ACK →
Comments	Establish a communication from network A to Network B, user B is not allocated in Network B. **Check:** Is a 410 Gone sent from Network B to Network A? Repeat this test in reverse direction.

Test case number	SS_unsucc_006
Test case group	BCALL/unsuccessful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	**The number of the called user is incomplete.** Ensure that the call will be released when the called number is incomplete. The network initiates call clearing to the calling user with 484 Not Found message.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 484 Address Incomplete ACK →
Comments	Establish a communication from network A to Network B, the called number is incomplete. **Check:** Is a 484 Address Incomplete sent from Network B to Network A? Repeat this test in reverse direction.

Test case number	SS_unsucc_007
Test case group	BCALL/unsuccessful
Reference	[3], [4] and [5]
SELECTION EXPRESSION	
Test purpose	**Session update requested by the calling user is unsuccessful, existing session remains unchanged.** 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 knows that it is to modify an existing session instead of establishing a new session. Ensure 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. The session remains unchanged.
Configuration	
SIP Parameter	INVITE: codec not supported in Network B

Message flow

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE	→	
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
		INVITE	→	
	←	488 Not Acceptable Here		
		ACK	→	
		Apply post test routine		

Comments	Establish a communication from network A to Network B. User A in Network A attempts to change the session by sending a SDP offer to the UE in Network B. Network B does not support the codec sent in the offer. **Check:** Is a 488 Not Acceptable Here sent from Network B to Network A? Repeat this test in reverse direction.

Test case number	SS_unsucc_008
Test case group	BCALL/unsuccessful
Reference	[3], [4] and [5]
SELECTION EXPRESSION	
Test purpose	**Session update requested by the called user is unsuccessful, existing session remains unchanged.** 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. Ensure 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. The session remains unchanged. The 488 Not Acceptable Here may be sent by a simulation equipment.
Configuration	
SIP Parameter	INVITE: codec not supported in Network A
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE	→	
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
	←	INVITE		
		488 Not Acceptable Here	→	
	←	ACK		
		Apply post test routine		

Comments	Establish a communication from network A to Network B. User B in Network B attempts to change the session by sending a SDP offer to the UE in Network A Network A does not support the codec sent in the offer. **Check:** Is a 488 Not Acceptable Here sent from Network B to Network A? Repeat this test in reverse direction.

Test case number	SS_unsucc_009
Test case group	BCALL/unsuccessful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	**Call clearing due to no answer from the called user initiated by the calling user.** Ensure that when there is no answer from the called user, the calling user initiates call clearing to the called user with CANCEL or BYE
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE	→	
	←	180 Ringing		
		CANCEL/BYE	→	
	←	200 OK CANCEL/BYE		
	←	487 Request Terminated		
		ACK	→	

Comments	**Check:** Is a CANCEL or BYE request is sent from the originating user? **Check:** Is a 487 Request Terminating send from the terminating user? **Check:** Are the media streams terminated after the 200 OK CANCEL/BYE was sent? Repeat this test in reverse direction.

Test case number	SS_unsucc_010
Test case group	BCALL/unsuccessful
Reference	[3], [4] and [5]
SELECTION EXPRESSION	
Test purpose	**Codec not supported by the called user.** The initial INVITE contains a SDP with codes that does not support by the called user. Ensure that, when the called user does not accept the Media session, the called user initiate call clearing to the calling user with 488 Not Acceptable Here, which also receives an ACK.
Configuration	
SIP Parameter	INVITE: codec not supported at user (Network B)
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** → INVITE → ← 488 Not Acceptable Here ← → ACK →
Comments	Establish a call setup from network A to Network B. User B in Network B does not support the codec offered in the SDP received from Network A. **Check:** Is a 488 Not Acceptable Here sent from Network B to Network A. Repeat this test in reverse direction.

Test case number	SS_unsucc_011
Test case group	BCALL/unsuccessful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	**Call clearing due to no answer from the called user initiated by the originating network.** Ensure that when there is no answer from the called user, the originating network initiate the call clearing after timeout of SIP timer C and sends a CANCEL or BYE to the called user.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** → INVITE → ← 180 Ringing **Start timer C** **Timeout timer C** CANCEL/BYE → ← 200 OK CANCEL/BYE ← 487 Request Terminated ACK →
Comments	**Check:** Is a CANCEL or BYE request is sent by the originating network? **Check:** Is a 487 Request Terminating send from the terminating user? **Check:** Are the media streams terminated after the 200 OK CANCEL/BYE was sent? Repeat this test in reverse direction.

Test case number	SS_unsucc_012
Test case group	BCALL/unsuccessful
Reference	6.11.2/[24]
SELECTION EXPRESSION	[Network B] SE 17
Test purpose	**SIP-I support. Called number is not allocated in the PSTN/PLMN network.** Ensure that, when calling to an unallocated number in the PSTN/PLMN part of network B and ISUP - SIP-I interworking applies in Network B, the network initiate call clearing to the calling user with a 404 Not Found message. A ISUP REL message is encapsulated and the **Cause value** indicator is set to '1'.
Configuration	The called user number is not assigned to the PSTN/PLMN part in Network B
SIP Parameter	404: Reason: Q.850;cause=1 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 1 --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 404 Not Found(REL) ACK →
Comments	Establish a communication from network A to Network B, called user number is not allocated in the PSTN/PLMN part of Network B **Check:** Is a 404 Not Found sent from Network B to Network A? **Check:** is a ISUP REL encapsulated and the **Cause value** indicator is set to '1'? **Check:** If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL? Repeat this test in reverse direction.

Test case number	SS_unsucc_013
Test case group	BCALL/unsuccessful
Reference	6.11.2/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	**SIP-I support. The called user is busy.** Ensure that, when the called user in the PSTN/PLMN part of Network B and ISUP - SIP-I interworking applies in Network B is busy, the network initiates call clearing to the calling user with a 486 Busy Here message. A ISUP REL message is encapsulated and the **Cause value** indicator is set to '17'.
Configuration	The called user is busy in the PSTN/PLMN part in Network B
SIP Parameter	486: Reason: Q.850;cause=17 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 --[any boundary name]--
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE → ← 486 Busy Here(REL) ACK →	
Comments	Establish a communication from network A to Network B, user B in the PSTN/PLMN part of Network B is busy. **Check:** Is a 486 Busy Here sent from Network B to Network A? **Check:** Is a ISUP REL encapsulated and the **Cause value** indicator is set to '17'? **Check:** If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_unsucc_014</td></tr>
<tr><td>Test case group</td><td>BCALL/unsuccessful</td></tr>
<tr><td>Reference</td><td>6.11.2/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network B] SE 17 AND SE 47</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. The called user rejects the call.

Ensure that, when the called user in the PSTN/PLMN part of Network B and ISUP - SIP-I interworking applies in Network B rejects the communication setup, the network initiates call clearing to the calling user with a 480 Temporarily Unavailable final response. A ISUP REL message is encapsulated and the Cause value indicator is set to '21'.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td>480:
Reason: Q.850;cause=21 (optional)
Content-Type: multipart/mixed;boundary=[any boundary name]

--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

REL
Cause value: 21

--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
INVITE →
← 480 Temporarily Unavailable (REL)
ACK →</td></tr>
<tr><td>Comments</td><td>Establish a communication from network A to Network B, user B in the PSTN/PLMN part of network B rejects the communication setup.
Check: Is a 480 Temporarily Unavailable sent from Network B to Network A?
Check: is a ISUP REL encapsulated and the Cause value indicator is set to '21'?
Check: If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_unsucc_015
Test case group	BCALL/unsuccessful
Reference	7.7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47
Test purpose	**SIP-I support. Call clearing due to no answer from the called user initiated by the calling user.** Ensure when the early dialogue is not confirmed by the called user, the calling user located in the PSTN/PLMN part of Network A and ISUP - SIP-I interworking applies in Network A initiates call clearing to the called user with CANCEL or BYE. An ISUP REL message is encapsulated in the BYE request and the **Cause value** indicator is set to '16'.
Configuration	
SIP Parameter	480: Reason: Q.850;cause=16 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 16 --[any boundary name]--
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE	→	
	←	180 Ringing		
CASE A				
		CANCEL	→	
	←	200 OK CANCEL		
	←	487 Request Terminated		
		ACK	→	
CASE B				
		BYE(REL)	→	
	←	200 OK BYE(RLC)		
	←	487 Request Terminated		
		ACK	→	

Comments	Establish a communication from network A to Network B, user B does not confirm the communication. The originating user in the PSTN/PLMN part of Network A terminates the early dialogue. **Check:** Is a CANCEL or BYE request is sent from the originating network? **Check:** Is a ISUP REL encapsulated in a BYE request? **Check:** Is the **Cause value** of the encapsulated REL set to '16'? **Check:** If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL? **Check:** Is a 487 Request Terminating send from the terminating user? **Check:** Are the media streams terminated after the 200 OK CANCEL/BYE was sent? **NOTE:** A ISUP REL is not encapsulated in a CANCEL request. Repeat this test in reverse direction.

Test case number	SS_unsucc_016
Test case group	BCALL/unsuccessful
Reference	7.7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47
Test purpose	**SIP-I support. Call clearing due to no answer from the called user initiated by the originating network.** Ensure when the early dialogue is not confirmed by the called user, the originating network initiate the call clearing after timeout of ISUP timer T9 if the calling user is located in the PSTN/PLMN part of Network A and ISUP - SIP-I interworking applies in Network A and the originating network sends a CANCEL or BYE to the called user. An ISUP REL message is encapsulated in the BYE request and the **Cause value** indicator is set to '19'.
Configuration	
SIP Parameter	480: Reason: Q.850;cause=19 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 19 --[any boundary name]--

Message flow

SIP (Network A)		Interconnection Interface		SIP (Network B)
	→	INVITE	→	
	←	180 Ringing		
		Start timer T9		
		Timeout T9		
CASE A				
		CANCEL	→	
	←	200 OK CANCEL		
	←	487 Request Terminated		
		ACK	→	
CASE B				
		BYE(REL)	→	
	←	200 OK BYE(RLC)		
	←	487 Request Terminated		
		ACK	→	

Comments	Establish a communication from network A to Network B, user B does not answer the communication setup. The ISUP timer T9 in the PSTN/PLMN expires **Check:** Is a CANCEL or BYE request is sent by the originating network? **Check:** Is a ISUP REL encapsulated in a BYE request? **Check:** Is the **Cause value** of the encapsulated REL set to '19'? **Check:** If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL? **Check:** Is a 487 Request Terminating send from the terminating user? **Check:** Are the media streams terminated after the 200 OK CANCEL/BYE was sent? **NOTE:** A ISUP REL is not encapsulated in a CANCEL request. Repeat this test in reverse direction.

7.1.5 Test purposes for Supplementary services

7.1.5.1 Test purposes for OIP

Test case number	SS_oip_001
Test case group	SIP-SIP/Service/OIP
Reference	5.2.6.3/[2]
SELECTION EXPRESSION	
Test purpose	**No P-Preferred-Identity received. The terminating user receives the default public user identity of the originating user.** In case the preconditions are fulfilled to provide the terminating UE with originating identification information without preventing the presentation, ensure that no identity information in the P-Preferred-Identity header 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.
Configuration	
SIP Parameter	INVITE P-Asserted-Identity= default public user identity
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE →
Comments	**Check:** Is the P-Asserted-Identity set to the default public user identity? **Check:** Is optional a second P-Asserted-Identity header present as a 'tel' URI with a public user identity? **Check:** Is the user parameter is set to phone? Repeat this test in reverse direction. Repeat this test with all relevant end devices.

Test case number	SS_oip_002
Test case group	SIP-SIP/Service/OIP
Reference	5.2.6.3/[2]
SELECTION EXPRESSION	
Test purpose	**P-Preferred-Identity received, no match with the set of registered public identities. The terminating user receives the default public user identity of the originating user.** In case the preconditions are fulfilled to provide the terminating UE with originating identification information without preventing the presentation, ensure that an identity information in the P-Preferred-Identity header 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.
Configuration	
SIP Parameter	INVITE P-Asserted-Identity= default public user identity
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE →
Comments	**Check:** Is the P-Asserted-Identity set to the default public user identity? **Check:** Is optional a second P-Asserted-Identity header present as a 'tel' URI with a public user identity? **Check:** Is the user parameter is set to phone? **Check:** Is the P-Preferred-Identity header not present? Repeat this test in reverse direction. Repeat this test with all relevantend devices.

Test case number	SS_oip_003
Test case group	SIP-SIP/Service/OIP
Reference	5.2.6.3/[2]
SELECTION EXPRESSION	
Test purpose	**P-Preferred-Identity received, match with the set of registered public identities. The terminating user receives the registered public user identity of the originating user.** In case the preconditions are fulfilled to provide the terminating UE with originating identification information without preventing the presentation, ensure that an identity information in the P-Preferred-Identity header 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.
Configuration	
SIP Parameter	INVITE P-Asserted-Identity= matched public user identity'
Message flow	**SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE →
Comments	**Check:** Is the P-Asserted-Identity set to the identified public user identity? **Check:** Is optional a second P-Asserted-Identity header present as a 'tel' URI with a public user identity? **Check:** Is the user parameter is set to phone? **Check:** Is the P-Preferred-Identity header not present? Repeat this test in reverse direction. Repeat this test with all relevantend devices.

Test case number	SS_oip_004
Test case group	SIP-SIP/Service/OIP
Reference	4.5.2.4/[7]
SELECTION EXPRESSION	SE 18 AND NOT SE 19
Test purpose	**No Special arrangement exists.** The special arrangement does not exist (screening of user provided information). The network compares the information in the From header with the set of registered public identities of the originating user If is no match is found, the AS sets the From header to the SIP URI that includes the registered default public user identity.
Configuration	Special arrangement for the originating user **does not** exist
SIP Parameter	INVITE From=default public user identity P-Asserted-Header=[any registered public user identity]
Message flow	**SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE →
Comments	**Check:** Is the From header URI set to the value of the P-Asserted-Identity URI? **Check:** Is the P-Asserted-Identity set to any registered public user identity? **Check:** Is the user parameter is set to phone? Repeat this test in reverse direction. Repeat this test with all relevantend devices.

Test case number	SS_oip_005
Test case group	SIP-SIP/Service/OIP
Reference	4.5.2.4/[7]
SELECTION EXPRESSION	SE 18 AND SE 19
Test purpose	**Special arrangement exists.** The special arrangement exists (no screening of user provided information). 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.
Configuration	Special arrangement for the originating user exists
SIP Parameter	INVITE From= original value P-Asserted-Header=[any registered public user identity]
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE →	
Comments	**Check:** Is the From header URI set to original value sent by the user? **Check:** Is the P-Asserted-Identity set to any registered public user identity? **Check:** Is the user parameter is set to phone? Repeat this test in reverse direction. Repeat this test with all relevantend devices.

Test case number	SS_oip_006
Test case group	SIP-SIP/Service/OIP
Reference	7.1.3/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 52
Test purpose	**SIP-I support. ISUP Calling party number *presentation allowed* in the encapsulated IAM.** Ensure when BICC/ISUP - SIP-I interworking applies in the originating network the BICC/ISUP IAM is encapsulated in the INVITE request. The P-Asserted-Identity header field is derived from the Calling party number in the encapsulated IAM. The 'Presentation restriction' indicator in the encapsulated IAM is set to 'allowed' no Privacy value 'id' is present in the INVITE request.
Configuration	
SIP Parameter	INVITE P-Asserted-Identity=[**derived from the ISUP calling party number**] Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** **Calling party number** Screening indicator Network provided or user provided, verified and passed Presentation restriction **allowed** Address signal --[any boundary name]--
Message flow SIP (Network A)	**Interconnection Interface** INVITE(IAM) → **SIP (Network B)**
Comments	**Check:** Is a BICC/ISUP IAM encapsulated in the in the INVITE request? **Check:** Is the **Calling party number** present in the encapsulated IAM and the screening indicator is set to 'Network provided' or 'user provided, verified and passed' and the Presentation restriction indicator is set to 'allowed'? **Check:** Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM? **Check:** Is the value 'id' not present in the Privacy header field (if included)? Repeat this test in reverse direction.

Test case number	SS_oip_007
Test case group	SIP-SIP/Service/OIP
Reference	7.1.3/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 52
Test purpose	**SIP-I support. ISUP Additional Calling party number *presentation allowed* in the encapsulated IAM.** Ensure when BICC/ISUP - SIP-I interworking applies in the originating network the BICC/ISUP IAM is encapsulated in the INVITE request. The From field is derived from the Additional Calling party number in the encapsulated IAM. The 'Presentation restriction' indicator in the encapsulated IAM is set to 'allowed' no Privacy value 'id' is present in the INVITE request.
Configuration	The originating user in the PSTN/PLMN part of Network A is subscribed to the 'no screening option'
SIP Parameter	INVITE From=[**derived from the ISUP Additional calling party number**] P-Asserted-Identity=[**derived from the ISUP calling party number**] Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Calling party number Screening indicator Network Provided **Presentation restriction** allowed **Address signal** **Generic number** Number Qualifier Indicator **Additional calling party number** Screening indicator user provided, not verified **Presentation restriction** allowed **Address signal** --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(IAM) →
Comments	**Check:** Is a BICC/ISUP IAM encapsulated in the in the INVITE request? **Check:** Is the **Calling party number** present in the encapsulated IAM and the screening indicator is set to 'Network Provided' and the Presentation restriction indicator is set to 'allowed'? **Check:** Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM? **Check:** Is a Generic number parameter, Number Qualifier Indicator set to **Additional calling party number** present and the screening indicator is set to 'user provided, not verified' and the Presentation restriction indicator is set to 'allowed'? **Check:** Is the From header field derived from the Additional calling party number in the encapsulated IAM? **Check:** Is the value 'id' not present in the Privacy header field (if included)? Repeat this test in reverse direction.

7.1.5.2 Test purposes for OIR

Test case number	SS_oir_001
Test case group	SIP-SIP/Service/OIR
Reference	4.3.2, 4.5.2.4/ [7]
SELECTION EXPRESSION	SE 20
Test purpose	**Terminating user does not receive the identity of the originating user.** In case the preconditions are fulfilled not to provide the terminating UE with originating identification information (e.g. permanent mode), ensure that the P-Asserted-Identity still contains identity information and the privacy is set to 'id' or 'header' or 'user'. The terminating user does not receive the identity of the originating user. As a network option, the From header is set to an anonymous User Identity.
Configuration	Originating user subscribes to the OIR service
SIP Parameter	INVITE P-Asserted-Identity: Privacy:**id** OR **header** OR **user** From: <sip:anonymous@anonymous.invalid> (optional)
Message flow **SIP (Network A)**	**Interconnection Interface** INVITE ➔ **SIP (Network B)**
Comments	**Check:** Is the P-Asserted-Identity is present? **Check:** Is the Privacy header set to 'id' or 'header' or 'user'? **Check:** Is optional the From header set to an anonymous User Identity? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_oir_002
Test case group	SIP-SIP/Service/OIR
Reference	4.3.2, 4.5.2.4/[7]
SELECTION EXPRESSION	SE 20 AND SE 25
Test purpose	**Communication forwarding unconditional, served user subscribes OIR.** The user A and user C are in network B and user C is provided with OIP. The user B is in network A and is provided with CFU "diverting number is released to the diverted-to user"=Yes. In case the served user subscribes Originating Identification Restriction (e.g. permanent mode), ensure that when user A calls user B, the call is forwarded unconditional to user C, user C is not informed of the forwarding number. The diverted-to user receives no identity of the diverting user neither in a History-Info header nor in the To header.
Configuration	Diverting user subscribes to the OIR service
SIP Parameter	INVITE: no history entry present INVITE: History-Info header: <sip:userB@networkA?Privacy=history >;index=1, <sip: userC@networkB;cause=302 >;index=1.1
Message flow **SIP (Network A)**	**Interconnection Interface** 🡐 INVITE **CFU is performed in Network A** INVITE ➔ **Apply post test routine** **SIP (Network B)**
Comments	**Check:** No History-Info header is received in the INVITE from Network B. **Check:** Is the Privacy value history is escaped in the hi-targed-to-uri of the diverting user in Network A? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_oir_003
Test case group	SIP-SIP/Service/OIR
Reference	7.1.3/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 52
Test purpose	**SIP-I support. ISUP Calling party number *presentation restricted* in the encapsulated IAM.** Ensure when BICC/ISUP - SIP-I interworking applies in the originating network the BICC/ISUP IAM is encapsulated in the INVITE request. The P-Asserted-Identity header field is derived from the Calling party number in the encapsulated IAM. The 'Presentation restriction' indicator in the encapsulated IAM is set to 'restricted' the value 'id' is present in the Privacy header of the INVITE request.
Configuration	
SIP Parameter	INVITE P-Asserted-Identity=[derived from the ISUP calling party number] Privacy: id Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Calling party number Screening indicator Network provided or user provided, verified and passed Presentation restriction restricted Address signal --[any boundary name]--
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE(IAM) →	
Comments	**Check:** Is a BICC/ISUP IAM encapsulated in the in the INVITE request? **Check:** Is the **Calling party number** present in the encapsulated IAM and the screening indicator is set to 'Network provided' or 'user provided, verified and passed' and the Presentation restriction indicator is set to 'restricted'? **Check:** Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM? **Check:** Is the value 'id' present in the Privacy header field? Repeat this test in reverse direction.

Test case number	SS_oir_004
Test case group	SIP-SIP/Service/OIR
Reference	7.1.3/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 52
Test purpose	**SIP-I support. ISUP Additional Calling party number *presentation restricted* in the encapsulated IAM.** Ensure when BICC/ISUP - SIP-I interworking applies in the originating network the BICC/ISUP IAM is encapsulated in the INVITE request. The From field is derived from the Additional Calling party number in the encapsulated IAM. The 'Presentation restriction' indicator in the Generic number parameter is set to 'allowed' no Privacy value 'id' is present in the INVITE request.
Configuration	The originating user in the PSTN/PLMN part of Network A is subscribed to the 'no screening option'
SIP Parameter	INVITE P-Asserted-Identity=[**derived from the ISUP calling party number**] From=[**derived from the ISUP Additional calling party number**] Privacy: **id** Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Calling party number Screening indicator Network Provided **Presentation restriction** restricted **Address signal** **Generic number** Number Qualifier Indicator **Additional calling party number** Screening indicator user provided, not verified **Presentation restriction** restricted **Address signal** --[any boundary name]--
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE(IAM) →	
Comments	**Check:** Is a BICC/ISUP IAM encapsulated in the in the INVITE request? **Check:** Is the **Calling party number** present in the encapsulated IAM and the screening indicator is set to 'Network Provided' and the Presentation restriction indicator is set to 'restricted'? **Check:** Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM? **Check:** Is a Generic number parameter, Number Qualifier Indicator set to **Additional calling party number** present and the screening indicator is set to 'user provided, not verified' and the Presentation restriction indicator is set to 'restricted'? **Check:** Is the From header field derived from the Additional calling party number in the encapsulated IAM? **Check:** Is the value 'id' present in the Privacy header field? Repeat this test in reverse direction.

7.1.5.3 Test purposes for TIP

Test case number	SS_tip_001
Test case group	SIP-SIP/Service/TIP
Reference	5.2.6.4/[8]
SELECTION EXPRESSION	
Test purpose	**Originating user receives the identity of the terminating user.** Ensure in case the preconditions are fulfilled to provide the originating UE with terminating identification information without preventing the presentation , the originating UE receives in a 1xx or 200 SIP response a **P-Asserted-Identity** header field with a valid public user identity of the terminating UE.
Configuration	
SIP Parameter	18x/200 OK INVITE P-Asserted-Identity:
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → **CASE A** ← 180 Ringing **CASE B** ← 183 Session Progress **CASE C** ← 200 OK INVITE(P-Asserted-Identity) **Apply post test routine**
Comments	**Check:** Is the P-Asserted-Identity is present in a 180 Ringing or 183 Session Progress or in a 200 OK INVITE? Repeat this test in reverse direction. Repeat this test with all relevant end devices.

Test case number	SS_tip_002
Test case group	SIP-SIP/Service/TIP
Reference	4.5.2.9/[8]
SELECTION EXPRESSION	SE 21 AND SE 22 AND [Network B] SE 48
Test purpose	**Second identity provided in UPDATE.** Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request and the terminating UE receives the from-change tag, The terminating user sends a 'from-change' tag in the supported header in the 200 OK INVITE a second identity is provided in the UPDATE request sent by the terminated user in the From header after the ACK was received.
Configuration	Special arrangement for the terminating user exists
SIP Parameter	INVITE Supported: from-change 200 OK INVITE P-Asserted-Identity: UPDATE From: (second user identity)
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE	→	
	←	180 Ringing		
	←	200 OK INVITE(P-Asserted-Identity)		
		ACK	→	
	←	UPDATE (From)		
		200 OK UPDATE	→	
		Apply post test routine		

Comments	**Check:** Is the 'from-change' tag present in the Supported header of the initial INVITE request? **Check:** Is the P-Asserted-Identity is present in a 180 Ringing or 183 Session Progress or in a 200 OK INVITE? **Check:** Is the 'from-change' tag present in the supported header of the provisional (18x) or final (200 OK) response? **Check:** Is an UPDATE request sent by the terminating user containing a From header field set to the value send by the terminating user? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_tip_003
Test case group	SIP-SIP/Service/TIP
Reference	4.5.2.9/[8]
SELECTION EXPRESSION	SE 21 AND SE 22 AND [Network B] SE 48
Test purpose	**Second identity not provided.** Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request, the terminating user does not receive the from-change tag in the initial INVITE, no from-change tag is sent in the 200 OK INVITE response, an UPDATE containing a second identity is sent and the From header is set to the default public user identity of the terminating user.
Configuration	Special arrangement for the terminating user **does not** exist
SIP Parameter	INVITE Supported: from-change 200 OK INVITE P-Asserted-Identity: UPDATE From: (default public user identity)
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing ← 200 OK INVITE(P-Asserted-Identity) ACK → ← UPDATE (From) 200 OK UPDATE → **Apply post test routine**
Comments	**Check:** Is the 'from-change' tag present in the Supported header of the initial INVITE request? **Check:** Is the P-Asserted-Identity is present in the 200 OK INVITE? **Check:** Is the 'from-change' tag present in the supported header of the provisional (18x) or final (200 OK) response? **Check:** Is an UPDATE request sent by the terminating user containing a From header field set to the public user identity of the terminating user? Repeat this test in reverse direction. Repeat this test with all relevant end devices.

Test case number	SS_tip_004
Test case group	SIP-SIP/Service/TIP
Reference	6.7/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. The Connected number presentation allowed is present in the encapsulated 200 OK.** Ensure that on receipt of a 200 OK INVITE to establish a confirmed dialogue an ANM is encapsulated if SIP-I - BICC/ISUP interworking is applicable in Network B. The Address presentation restriction indicator is set to 'allowed'. The screening indicator is set to Network provided or user provided, verified and passed.
Configuration	
SIP Parameter	200 OK INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** **Connected number** Screening indicator Network provided or user provided, verified and passed Address presentation restriction **allowed** Address signal --[any boundary name]--
Message flow	**SIP (Network A)** — **Interconnection Interface** — **SIP (Network B)** INVITE(IAM) → ← 180 Ringing(ACM) ← 200 OK INVITE(**ANM**) ACK → **Apply post test routine**
Comments	**Check:** Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response? **Check:** Is the Screening indicator in the encapsulated ANM set to 'Network provided' or 'user provided, verified and passed'? **Check:** Is the Address presentation restriction indicator in the encapsulated ANM set to allowed? Repeat this test in reverse direction.

Test case number	SS_tip_005
Test case group	SIP-SIP/Service/TIP
Reference	6.7/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. The additional connected number restricted is present in the encapsulated 200 OK.** Ensure that on receipt of a 200 OK INVITE to establish a confirmed dialogue an ANM is encapsulated if SIP-I - BICC/ISUP interworking is applicable in Network B. A Generic number parameter is present the Number qualifier indicator set to 'additional connected number' the Screening indicator is set to 'user provided, not verified' and the Address Presentation Restricted is set to 'allowed'. A Connected number parameter is present the Screening indicator is set to 'Network provided' and the Address Presentation Restricted indicator is set to 'allowed'.
Configuration	The terminating user in the PSTN/PLMN part of Network B is subscribed to the COLP 'no screening option'
SIP Parameter	200 OK INVITE P-Asserted-Identity=[**derived from the ISUP Connected number**] Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** **Connected number** Screening indicator Network provided or user provided, verified and passed Presentation restriction **allowed** Address signal **Generic number** Number Qualifier Indicator **Additional calling party number** Screening indicator user provided, not verified Address Presentation Restricted allowed Address signal --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(IAM) → ← 180 Ringing(ACM) ← 200 OK INVITE(**ANM**) ACK → **Apply post test routine**
Comments	**Check:** Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response? **Check:** Is a Generic number parameter present in the encapsulated ANM? **Check:** Is the Number Qualifier Indicator of the Generic number set to 'additional connected number'? **Check:** Is the Screening indicator of the Generic number set to 'user provided, not verified'? **Check:** Is the Address presentation restriction indicator in the Generic number set to 'allowed'? Repeat this test in reverse direction.

7.1.5.4 Test purposes for TIR

Test case number	SS_tir_001		
Test case group	SIP-SIP/Service/TIR		
Reference	4.5.2.9/[8]		
SELECTION EXPRESSION	SE 23		
Test purpose	**Originating user does not receive the identity of the terminating user.** Ensure that, when the preconditions are fulfilled to prevent the presentation of the terminating user identity at the originating user, the originating UE receives, in any non-100 SIP response (e.g. 180, 183, 200), a Privacy header field is set to "id" and no P-Asserted-Identity header field is present.		
Configuration	The terminating user subscribes to the 'TIR' service		
SIP Parameter	18x/200 OK INVITE P-Asserted-Identity: Privacy: id		
Message flow			
SIP (Network A)		**Interconnection Interface**	**SIP (Network B)**
		INVITE →	
CASE A	←	180 Ringing	
CASE B	←	183 Session Progress	
CASE C	←	200 OK INVITE(P-Asserted-Identity)	
		Apply post test routine	
Comments	**Check:** Is the P-Asserted-Identity is present in the provisional (18x) or final (200 OK) response? **Check:** Is the Privacy header in the provisional (18x) or final (200 OK) response is set to 'id'? Repeat this test in reverse direction. Repeat this test with all chosen end devices.		

Test case number	SS_tir_002
Test case group	SIP-SIP/Service/TIR
Reference	6.7/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. The Connected number presentation allowed is present in the encapsulated 200 OK.** Ensure that on receipt of a 200 OK INVITE to establish a confirmed dialogue an ANM is encapsulated if SIP-I - BICC/ISUP interworking is applicable in Network B. The Address presentation restriction indicator is set to 'restricted'. The screening indicator is set to 'Network provided' or 'user provided, verified and passed'.
Configuration	
SIP Parameter	200 OK INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** **Connected number** Screening indicator Network provided or user provided, verified and passed Address presentation restriction restricted Address signal --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(IAM) → ← 180 Ringing(ACM) ← 200 OK INVITE(**ANM**) ACK → **Apply post test routine**
Comments	**Check:** Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response? **Check:** Is the Screening indicator in the encapsulated ANM set to 'Network provided' or 'user provided, verified and passed'? **Check:** Is the Address presentation restriction indicator in the encapsulated ANM set to allowed? Repeat this test in reverse direction.

Test case number	SS_tir_003
Test case group	SIP-SIP/Service/TIR
Reference	6.7/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. The additional connected number restricted is present in the encapsulated 200 OK.** Ensure that on receipt of a 200 OK INVITE to establish a confirmed dialogue an ANM is encapsulated if SIP-I - BICC/ISUP interworking is applicable in Network B. A Generic number parameter is present the Number qualifier indicator set to 'additional connected number' the Screening indicator is set to 'user provided, not verified' and the Address Presentation Restricted is set to 'restricted'. A Connected number parameter is present the Screening indicator is set to 'Network provided' and the Address Presentation Restricted indicator is set to 'restricted'.
Configuration	The terminating user in the PSTN/PLMN part of Network B is subscribed to the COLP 'no screening option'
SIP Parameter	200 OK INVITE P-Asserted-Identity=[**derived from the ISUP Connected number**] Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** **Connected number** Screening indicator Network provided or user provided, verified and passed Presentation restriction restricted Address signal **Generic number** Number Qualifier Indicator **Additional calling party number** Screening indicator user provided, not verified Address Presentation Restricted restricted Address signal --[any boundary name]--
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(IAM)	→	
	←	180 Ringing(ACM)		
	←	200 OK INVITE(**ANM**)		
		ACK	→	
		Apply post test routine		

Comments	**Check:** Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response? **Check:** Is a Generic number parameter present in the encapsulated ANM? **Check:** Is the Number Qualifier Indicator of the Generic number set to 'additional connected number'? **Check:** Is the Screening indicator of the Generic number set to 'user provided, not verified'? **Check:** Is the Address presentation restriction indicator in the Generic number set to 'allowed'? Repeat this test in reverse direction.

7.1.5.5 Communication Hold (HOLD)

Test case number	SS_hold_001
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[17]
SELECTION EXPRESSION	SE 24
Test purpose	**Hold the session the media stream was previously set to sendrecv.** Ensure that the UE A requesting hold of the session sends an INVITE or UPDATE request to hold the session. Hold is done containing the SDP with the attribute "a=sendonly". The UE A after requesting the hold session *receives* 200 OK final response containing the SDP with the attribute "a=recvonly".
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		A confirmed session already exists		
CASE A		INVITE(**sendonly**)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
CASE B		UPDATE(**sendonly**)	→	
	←	200 OK UPDATE (**recvonly**)		
		Apply post test routine		

Comments	**Check:** Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? Repeat this test in reverse direction.

Test case number	SS_hold_002
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[17]
SELECTION EXPRESSION	SE 24
Test purpose	**Hold the session the media stream was previously set to recvonly.** Ensure that the UE A requesting hold of the session stops sending media and sends an INVITE or UPDATE request to hold the session. Hold is done containing the SDP with the attribute "a=inactive". The UE A after requesting to resume the held session *receives* 200 OK final response containing the SDP with the attribute "a=inactive."
Configuration	
SIP Parameter	

Message flow

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		A confirmed session already exists		
CASE A	←	INVITE (**sendonly**)		
		200 OK INVITE (recvonly)	→	
	←	ACK		
		INVITE (**inactive**)	→	
	←	200 OK INVITE (inactive)		
		ACK	→	
CASE B	←	INVITE (**sendonly**)		
		200 OK INVITE (recvonly)	→	
	←	ACK		
		UPDATE(**inactive**)	→	
	←	200 OK UPDATE (inactive)		
CASE C	←	UPDATE (**sendonly**)		
		200 OK UPDATE (recvonly)	→	
		INVITE (**inactive**)	→	
	←	200 OK INVITE (inactive)		
		ACK	→	
CASE D	←	UPDATE (**sendonly**)		
		200 OK UPDATE (recvonly)	→	
		UPDATE(**inactive**)	→	
	←	200 OK UPDATE (inactive)		
		Apply post test routine		

Comments	**Check:** Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? Repeat this test in reverse direction.

Test case number	SS_hold_003
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[17]
SELECTION EXPRESSION	SE 24
Test purpose	**Resume the session the media stream was previously set to sendonly.** Ensure that the UE A is requested to resume the session with user B the UE-A starts sending media and sends an INVITE or UPDATE request to resume the session with the attribute "a=sendrecv in the SDP. The UE A after requesting to resume the held 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.
Configuration	
SIP Parameter	
Message flow	
Comments	**Check:** Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network A able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? The absence of the 'sendrecv' attribute is the default value. Repeat this test in reverse direction.

Message flow:

SIP (Network A)		Interconnection Interface		SIP (Network B)
		A confirmed session already exists		
CASE A		INVITE (**sendonly**)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
		INVITE (**sendrecv**)	→	
	←	200 OK INVITE (sendrecv)		
		ACK	→	
CASE B		INVITE (**sendonly**)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
		UPDATE (**sendrecv**)	→	
	←	200 OK UPDATE (sendrecv)		
CASE C		UPDATE (**sendonly**)	→	
	←	200 OK UPDATE (recvonly)		
		INVITE (**sendrecv**)	→	
	←	200 OK INVITE (sendrecv)		
		ACK	→	
CASE D		UPDATE (**sendonly**)	→	
	←	200 OK UPDATE (recvonly)		
		UPDATE (**sendrecv**)	→	
	←	200 OK UPDATE (sendrecv)		
		Apply post test routine		

Test case number	SS_hold_004
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[17]
SELECTION EXPRESSION	SE 24
Test purpose	**Resume the session the media stream was previously set to inactive.** The Session is in the "inactive" state. Ensure that the UE A is requesting to resume the session with user B the UE-A sends an INVITE or UPDATE to resume the session with the attribute "a=recvonly in the SDP. The UE A after requesting to resume the held session *receives* 200 OK final response and optionally the attribute "a=sendonly in the SDP.
Configuration	
SIP Parameter	

Message flow

SIP (Network A)		Interconnection Interface		SIP (Network B)
		A confirmed session already exists		
CASE A	←	INVITE(sendonly)		
		200 OK INVITE (recvonly)	→	
	←	ACK		
		INVITE(**inactive**)	→	
	←	200 OK INVITE (inactive)		
		ACK	→	
		INVITE (**recvonly**)	→	
	←	200 OK INVITE (sendonly)		
		ACK	→	
CASE B	←	INVITE(sendonly)		
		200 OK INVITE (recvonly)	→	
	←	ACK		
		UPDATE(**inactive**)	→	
	←	200 OK UPDATE (inactive)		
		INVITE (**recvonly**)	→	
	←	200 OK INVITE (sendonly)		
		ACK	→	
CASE C	←	UPDATE (sendonly)		
		200 OK UPDATE (recvonly)	→	
		INVITE(**inactive**)	→	
	←	200 OK INVITE (inactive)		
		ACK	→	
		UPDATE (**recvonly**)	→	
	←	200 OK UPDATE (sendonly)		
CASE D	←	UPDATE (sendonly)		
		200 OK UPDATE (recvonly)	→	
		UPDATE(**inactive**)	→	
	←	200 OK UPDATE (inactive)		
		UPDATE (**recvonly**)	→	
	←	200 OK UPDATE (sendonly)		
		Apply post test routine		

Comments	**Check:** Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network A able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_hold_005</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/HOLD</td></tr>
<tr><td>Reference</td><td>4.5.2.1/[17]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>SE 24</td></tr>
<tr><td>Test purpose</td><td>Hold the session the media stream was previously set to sendrecv.

Ensure that the UE A receives an INVITE or UPDATE request to hold the session and stops sending media. Hold is done containing the SDP with the attribute "a=sendonly". The UE A after resuming the held session sends a 200 OK final response containing the SDP with the attribute "a=recvonly".</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td></td></tr>
<tr><td colspan="2">Message flow
SIP (Network A)   Interconnection Interface   SIP (Network B)
A confirmed session already exists
CASE A
← INVITE(sendonly)
200 OK INVITE(recvonly) →
← ACK

CASE B
← UPDATE(sendonly)
200 OK UPDATE (recvonly) →
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Check: Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_hold_006
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[17]
SELECTION EXPRESSION	SE 24
Test purpose	**Hold the session the media stream was previously set to sendonly.** The Session is in the "sendonly" state. Ensure that the UE A receives an INVITE or UPDATE request to hold the session and stops sending media. Hold is done containing the SDP with the attribute "a=inactive". The UE A after receiving the hold session *sends* 200 OK final response containing the SDP with the attribute "a=inactive".
Configuration	
SIP Parameter	

Message flow

SIP (Network A)		Interconnection Interface		SIP (Network B)
		A confirmed session already exists		
CASE A		INVITE(**sendonly**)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	INVITE (inactive)		
		200 OK INVITE (**inactive**)	→	
	←	ACK		
CASE B		INVITE(**sendonly**)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	UPDATE (inactive)		
		200 OK UPDATE (**inactive**)	→	
CASE C		UPDATE (**sendonly**)	→	
	←	200 OK UPDATE (recvonly)		
	←	INVITE (inactive)		
		200 OK INVITE (**inactive**)	→	
	←	ACK		
CASE D		UPDATE (**sendonly**)	→	
	←	200 OK UPDATE (recvonly)		
	←	UPDATE (inactive)		
		200 OK UPDATE (**inactive**)	→	
		Apply post test routine		

Comments	**Check:** Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request? **Check:** Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? Repeat this test in reverse direction.

Test case number	SS_hold_007
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[17]
SELECTION EXPRESSION	SE 24
Test purpose	**Resume the session the media stream was previously set to recvonly.** Ensure that the UE A receives an INVITE or UPDATE request requesting to resume the session with user B, the UE-A starts sending media. Resume is done containing the SDP with the attribute "a=sendrecv". The UE A after receiving the Resume of the session *sends* 200 OK final response containing the SDP with the attribute "a=sendrecv". The a=sendrecv attribute is the default value therefore the attribute can be omitted.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **A confirmed session already exists** **CASE A** ← INVITE (sendonly) 200 OK INVITE(**recvonly**) → ← ACK ← INVITE(sendrecv) 200 OK INVITE(**sendrecv**) → ← ACK **CASE B** ← UPDATE (sendonly) 200 OK UPDATE (**recvonly**) → ← UPDATE (sendrecv) 200 OK UPDATE (**sendrecv**) → **Apply post test routine**	
Comments	**Check:** Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network B able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? Repeat this test in reverse direction.

Test case number	SS_hold_008
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[17]
SELECTION EXPRESSION	SE 24
Test purpose	**Resume the session the media stream was previously set to inactive.** The Session is in the "inactive" state. Ensure that the UE A receives an INVITE or UPDATE request requesting to resume the session with user B, the UE-A starts sending media. Resume is done containing the SDP with the attribute "a=recvonly". The UE A after receiving the Resume of the session *sends* 200 OK final response containing the SDP with the attribute "a=sendonly". The a=sendrecv attribute is the default value therefore the attribute can be omitted.
Configuration	
SIP Parameter	

Message flow

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		A confirmed session already exists		
CASE A		INVITE (sendonly)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	INVITE (inactive)		
		200 OK INVITE (inactive)	→	
	←	ACK		
	←	INVITE (**recvonly**)		
		200 OK INVITE (sendonly)	→	
	←	ACK		
CASE B		INVITE (sendonly)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	UPDATE (inactive)		
		200 OK UPDATE (inactive)	→	
	←	UPDATE (**recvonly**)		
		200 OK UPDATE (sendonly)	→	
CASE C		UPDATE (sendonly)	→	
	←	200 OK UPDATE (recvonly)		
	←	INVITE (inactive)		
		200 OK INVITE (inactive)	→	
	←	ACK		
	←	INVITE (**recvonly**)		
		200 OK INVITE (sendonly)	→	
	←	ACK		
CASE D		UPDATE (sendonly)	→	
	←	200 OK UPDATE (recvonly)		
	←	UPDATE (inactive)		
		200 OK UPDATE (inactive)	→	
	←	UPDATE (**recvonly**)		
		200 OK UPDATE (sendonly)	→	
		Apply post test routine		

Comments	**Check:** Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network B able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? Repeat this test in reverse direction.

Test case number	SS_hold_009
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[17]
SELECTION EXPRESSION	SE 24
Test purpose	**Resume the session on both sides the media stream was previously set to inactive.** The Session is in the "inactive" state. Ensure that the UE A is requesting to resume the session with user B, the UE-A starts sending media and sends an INVITE or UPDATE request to resume the session with the attribute "a=**sendonly** in the SDP. The UE A after requests to resume the session *receives* 200 OK final response containing the SDP with the attribute "a=**recvonly**. The UE B after requests to resume the session *receives* 200 OK final response containing the SDP with the attribute "a=**sendrecv"**. The a=sendrecv attribute is the default value therefore the attribute can be omitted.

Message flow

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		A confirmed session already exists		
CASE A		INVITE(sendonly)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	INVITE(inactive)		
		200 OK INVITE (inactive)	→	
	←	ACK		
		INVITE(**sendonly**)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	INVITE(**sendrecv**)		
		200 OK INVITE (sendrecv)	→	
	←	ACK		
CASE B		INVITE(sendonly)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	UPDATE (inactive)		
		200 OK UPDATE (inactive)	→	
		INVITE(**sendonly**)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	UPDATE (**sendrecv**)		
		200 OK UPDATE (sendrecv)	→	
CASE C		UPDATE (sendonly)	→	
	←	200 OK UPDATE (recvonly)		
	←	INVITE(inactive)		
		200 OK INVITE (inactive)	→	
	←	ACK		
		UPDATE (**sendonly**)	→	
	←	200 OK UPDATE (recvonly)		
		ACK	→	
	←	INVITE(**sendrecv**)		
		200 OK INVITE (sendrecv)	→	
	←	ACK		
CASE D		UPDATE (sendonly)	→	
	←	200 OK UPDATE (recvonly)		
	←	UPDATE (inactive)		
		200 OK UPDATE (inactive)	→	
		UPDATE (**sendonly**)	→	
	←	200 OK UPDATE (recvonly)		
	←	UPDATE (**sendrecv**)		
		200 OK UPDATE (sendrecv)	→	
		Apply post test routine		

Comments	**Check:** Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network A able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network B able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? The absence of the 'sendrecv' attribute is the default value. Repeat this test in reverse direction.

Test case number	SS_hold_010
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[17]
SELECTION EXPRESSION	SE 24
Test purpose	**Resume the session on both sides the media stream was previously set to inactive.** The Session is in the "inactive" state. Ensure that the UE A receives an INVITE or UPDATE to resume the session with user B, the UE-A starts sending media. Resume is done containing the SDP with the attribute "a=recvonly". The UE A after receiving the Resume of the session *sends* 200 OK final response containing the SDP with the attribute "a=sendonly". The UE A after requests to resume the session *receives* 200 OK final response containing the SDP with the attribute "a=**sendrecv**. The UE B after receiving the Resume of the session *sends* 200 OK final response containing the SDP with the attribute "a=sendrecv". The a=sendrecv attribute is the default value therefore the attribute can be omitted.
Configuration	
SIP Parameter	

Message flow

SIP (Network A)		Interconnection Interface		SIP (Network B)
		A confirmed session already exists		
CASE A	←	INVITE(sendonly)		
		200 OK INVITE (recvonly)	→	
	←	ACK		
		INVITE(inactive)	→	
	←	200 OK INVITE (inactive)		
		ACK	→	
	←	INVITE(**sendonly**)		
		200 OK INVITE (recvonly)	→	
	←	ACK		
		INVITE(**sendrecv**)	→	
	←	200 OK INVITE (sendrecv)		
		ACK	→	
CASE B	←	INVITE(sendonly)		
		200 OK INVITE (recvonly)	→	
	←	ACK		
		UPDATE (inactive)	→	
	←	200 OK UPDATE (inactive)		
	←	INVITE(**sendonly**)		
		200 OK INVITE (recvonly)	→	
	←	ACK		
		UPDATE (**sendrecv**)	→	
	←	200 OK UPDATE (sendrecv)		
CASE C	←	UPDATE (sendonly)		
		200 OK UPDATE (recvonly)	→	
		INVITE(inactive)	→	
	←	200 OK INVITE (inactive)		
		ACK	→	
	←	UPDATE (**sendonly**)		
		200 OK UPDATE (recvonly)	→	
		INVITE(**sendrecv**)	→	
	←	200 OK INVITE (sendrecv)		
		ACK	→	
CASE D	←	UPDATE (sendonly)		
		200 OK UPDATE (recvonly)	→	
		UPDATE (inactive)	→	
	←	200 OK UPDATE (inactive)		
	←	UPDATE (**sendonly**)		
		200 OK UPDATE (recvonly)	→	
		UPDATE (**sendrecv**)	→	
	←	200 OK UPDATE (sendrecv)		
		Apply post test routine		

Comments	**Check:** Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network B able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? **Check:** Is the user in network A able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? The absence of the 'sendrecv' attribute is the default value. Repeat this test in reverse direction.

Test case number	SS_hold_011
Test case group	SIP-SIP/Service/HOLD
Reference	B.10/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 54
Test purpose	**SIP-I support. Hold requested by the calling user.** Ensure that when an INVITE request updates a confirmed session a CPG is encapsulated if ISUP - SIP-I interworking is applicable in Network A. The Generic Notification Indicator parameter is present set to 'hold'. The 'a' attribute is set to 'sendonly' present in the SDP. In the 200 OK INVITE the 'a' attribute is set to 'recvonly' present in the SDP.
Configuration	
SIP Parameter	NVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] a=sendonly --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic notification remote hold --[any boundary name]--
Message flow **SIP (Network A)** **CASE A**	**Interconnection Interface** **SIP (Network B)** **A confirmed session already exists** INVITE(sendonly, CPG hold) → ← 200 OK INVITE (recvonly) ACK → **Apply post test routine**
Comments	Establish a session from Network A to Network B The user in the PSTN/PLMN part of Network A places the session on hold. **Check:** Is a CPG encapsulated in the INVITE request? **Check:** Is a Generic notification parameter present the Notification indicator set to '**remote hold**'? **Check:** Is the 'a' attribute in the SDP set to '**sendonly**'? **Check:** Is the Version parameter in the SDP incremented? Repeat this test in reverse direction.

Test case number	SS_hold_012
Test case group	SIP-SIP/Service/HOLD
Reference	B.10/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 54
Test purpose	**SIP-I support. Hold requested by the called user.** Ensure that when an INVITE request updates a confirmed session a CPG is encapsulated if SIP-I - ISUP interworking is applicable in Network B. The Generic Notification Indicator parameter is present set to 'hold'. The 'a' attribute is set to 'sendonly' present in the SDP. In the 200 OK INVITE the 'a' attribute is set to 'recvonly' present in the SDP.
Configuration	
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] a=sendonly --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic notification remote hold --[any boundary name]--
Message flow **SIP (Network A)** **CASE A**	**Interconnection Interface** **SIP (Network B)** **A confirmed session already exists** ← INVITE(sendonly, CPG hold) 200 OK INVITE (recvonly) → ← ACK **Apply post test routine**
Comments	Establish a session from Network A to Network B The user in the PSTN/PLMN part of Network B places the session on hold. **Check:** Is a CPG encapsulated in the INVITE request? **Check:** Is a Generic notification parameter present the Notification indicator set to '**remote hold**'? **Check:** Is the 'a' attribute in the SDP set to '**sendonly**'? **Check:** Is the Version parameter in the SDP incremented? Repeat this test in reverse direction.

Test case number	SS_hold_013
Test case group	SIP-SIP/Service/HOLD
Reference	B.10/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 54
Test purpose	**SIP-I support. Hold requested by the originating user, Hold by the terminating user. Retrieve requested by the originating user.** Ensure the hold and retrieve procedure when ISUP - SIP-I interworking applies in the Network A: • Originating user in Network A places the session on hold. • Terminating user in Network B places the session on hold. • Originating user in Network A retrieves the session. • Terminating user in Network B retrieves the session. Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A.
Configuration	
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic notification remote hold or remote retrieval --[any boundary name]--
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		A confirmed session already exists		
CASE A		INVITE(sendonly, CPG hold)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	INVITE(inactive)		
		200 OK INVITE (inactive)	→	
	←	ACK		
		INVITE(sendonly, CPG retrieval)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	INVITE(sendrecv)		
		200 OK INVITE (sendrecv)	→	
	←	ACK		
		Apply post test routine		

Comments	Establish a session from Network A to Network B The user in the PSTN/PLMN part of Network A places the session on hold. **Check:** Is a CPG encapsulated in the INVITE request? **Check:** Is a Generic notification parameter present the Notification indicator set to '**remote hold**'? **Check:** Is the 'a' attribute in the SDP set to '**sendonly**'? **Check:** Is the Version parameter in the SDP incremented? The user in Network B places the session on hold **Check:** Is the 'a' attribute in the SDP set to '**inactive**'? **Check:** Is the Version parameter in the SDP incremented? The user in Network A retrieves the session **Check:** Is a CPG encapsulated in the INVITE request? **Check:** Is a Generic notification parameter present the Notification indicator set to '**remote retrieval**'? **Check:** Is the 'a' attribute in the SDP set to '**sendonly**'? **Check:** Is the Version parameter in the SDP incremented? The user in Network B retrieves the session **Check:** Is the 'a' attribute in the SDP set to '**sendrecv**'? **Check:** Is the Version parameter in the SDP incremented? Repeat this test in reverse direction.

Test case number	SS_hold_014
Test case group	SIP-SIP/Service/HOLD
Reference	B.10/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 54
Test purpose	**SIP-I support. Hold requested by the originating user, Hold by the terminating user. Retrieve requested by the terminating user.** Ensure the hold and retrieve procedure when ISUP - SIP-I interworking applies in the Network A: • Originating user in Network A places the session on hold. • Terminating user in Network B places the session on hold. • Terminating user in Network B retrieves the session. • Originating user in Network A retrieves the session. Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A.
Configuration	
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic notification remote hold or remote retrieval --[any boundary name]--
Message flow	
SIP (Network A)	**Interconnection Interface** / **SIP (Network B)** **A confirmed session already exists** INVITE(sendonly, CPG hold) → ← 200 OK INVITE (recvonly) ACK → ← INVITE(inactive) 200 OK INVITE (inactive) → ← ACK ← INVITE(recvonly) 200 OK INVITE (sendonly) → ← ACK INVITE(sendrecv, CPG retrieval) → ← 200 OK INVITE (sendrecv) ACK → **Apply post test routine**
Comments	Establish a session from Network A to Network B The user in the PSTN/PLMN part of Network A places the session on hold. **Check:** Is a CPG encapsulated in the INVITE request? **Check:** Is a Generic notification parameter present the Notification indicator set to '**remote hold**'? **Check:** Is the 'a' attribute in the SDP set to '**sendonly**'? **Check:** Is the Version parameter in the SDP incremented? The user in Network B places the session on hold **Check:** Is the 'a' attribute in the SDP set to '**inactive**'? **Check:** Is the Version parameter in the SDP incremented? The user in Network B retrieves the session **Check:** Is the 'a' attribute in the SDP set to '**recvonly**'? **Check:** Is the Version parameter in the SDP incremented? The user in Network A retrieves the session **Check:** Is a CPG encapsulated in the INVITE request? **Check:** Is a Generic notification parameter present the Notification indicator set to '**remote retrieval**'? **Check:** Is the 'a' attribute in the SDP set to '**sendrecv**'? **Check:** Is the Version parameter in the SDP incremented? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_hold_015</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/HOLD</td></tr>
<tr><td>Reference</td><td>B.10/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 17 AND SE 47 AND SE 54</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. Hold requested by the terminating user, Hold by the originating user. Retrieve requested by the originating user.

Ensure the hold and retrieve procedure when ISUP - SIP-I interworking applies in the Network A:
• Terminating user in Network B places the session on hold.
• Originating user in Network A places the session on hold.
• Originating user in Network A retrieves the session.
• Terminating user in Network B retrieves the session.
Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td>INVITE:
Content-Type: multipart/mixed;boundary=[any boundary name]

--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
CPG
Generic notification
remote hold
or
remote retrieval
--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
A confirmed session already exists
← INVITE(sendonly)
200 OK INVITE (recvonly) →
← ACK

INVITE(inactive, CPG hold) →
← 200 OK INVITE (inactive)
ACK →

INVITE(recvonly, CPG retrieval) →
← 200 OK INVITE (sendonly)
ACK →

← INVITE(sendrecv)
200 OK INVITE (sendrecv) →
← ACK
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Establish a session from Network A to Network B
The user in Network B places the session on hold.
Check: Is the 'a' attribute in the SDP set to 'sendonly'?
Check: Is the Version parameter in the SDP incremented?
The user in Network A places the session on hold
Check: Is a CPG encapsulated in the INVITE request?
Check: Is a Generic notification parameter present the Notification indicator set to 'remote hold'?
Check: Is the 'a' attribute in the SDP set to 'inactive'?
Check: Is the Version parameter in the SDP incremented?
The user in Network A retrieves the session
Check: Is a CPG encapsulated in the INVITE request?
Check: Is a Generic notification parameter present the Notification indicator set to 'remote retrieval'?
Check: Is the 'a' attribute in the SDP set to 'recvonly'?
Check: Is the Version parameter in the SDP incremented?
The user in Network B retrieves the session
Check: Is the 'a' attribute in the SDP set to 'sendrecv'?
Check: Is the Version parameter in the SDP incremented?
Repeat this test in reverse direction.</td></tr>
</table>

<table>
<tr><td>Test case number</td><td>SS_hold_016</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/HOLD</td></tr>
<tr><td>Reference</td><td>B.10/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 17 AND SE 47 AND SE 54</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. Hold requested by the terminating user, Hold by the originating user. Retrieve requested by the terminating user.

Ensure the hold and retrieve procedure when ISUP - SIP-I interworking applies in the Network A:
• Terminating user in Network B places the session on hold.
• Originating user in Network A places the session on hold.
• Terminating user in Network B retrieves the session.
• Originating user in Network A retrieves the session.
Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td>INVITE:
Content-Type: multipart/mixed;boundary=[any boundary name]

--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
CPG
Generic notification
remote hold
or
remote retrieval
--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
A confirmed session already exists
← INVITE(sendonly)
200 OK INVITE (recvonly) →
← ACK

INVITE(inactive, CPG hold) →
← 200 OK INVITE (inactive)
ACK →

← INVITE(sendonly)
200 OK INVITE (recvonly) →
← ACK

INVITE(sendrecv, CPG retrieval) →
← 200 OK INVITE (sendrecv)
ACK →
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Establish a session from Network A to Network B
The user in Network B places the session on hold.
Check: Is the 'a' attribute in the SDP set to 'sendonly'?
Check: Is the Version parameter in the SDP incremented?
The user in Network A places the session on hold
Check: Is a CPG encapsulated in the INVITE request?
Check: Is a Generic notification parameter present the Notification indicator set to 'remote hold'?
Check: Is the 'a' attribute in the SDP set to 'inactive'?
Check: Is the Version parameter in the SDP incremented?
The user in Network B retrieves the session
Check: Is the 'a' attribute in the SDP set to 'sendonly'?
Check: Is the Version parameter in the SDP incremented?
The user in Network A retrieves the session
Check: Is a CPG encapsulated in the INVITE request?
Check: Is a Generic notification parameter present the Notification indicator set to 'remote retrieval'?
Check: Is the 'a' attribute in the SDP set to 'sendrecv'?
Check: Is the Version parameter in the SDP incremented?
Repeat this test in reverse direction.</td></tr>
</table>

7.1.5.6 Communication Diversion (CDIV)

7.1.5.6.1 Communication Forwarding Unconditional (CFU)

Test case number	SS_cfu_001
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 25
Test purpose	**Communication forwarding unconditional, basic rules.** The user A and user C are in Network A. The user B is in network B and is provided with CFU. Ensure that when user A calls user B, the call is forwarded unconditional to user C. In the active call state, ensure the property of speech.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

Comments	**Check:** CDIV unconditional is successful. **Check:** In the active call state, ensure the property of speech. **Check:** Is the P-Asserted-Identity present set to the identity of the originating user? Repeat this test in reverse direction.

Test case number	SS_cfu_002
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	**Communication forwarding unconditional, no notification.** The user A and user C are in Network A. The user B is in network B and is provided with CFU, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B, the call is forwarded unconditional to user C, the originating user is not notified.
Configuration	**Subscription options:** Originating user receives notification that his communication has been diverted = No
SIP Parameter	
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		Apply post test routine		

Comments	**Check:** No notification regarding call forwarding in network B is received at the interconnection interface. Repeat this test in reverse direction.

Test case number	SS_cfu_003
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	**Communication forwarding unconditional, originating user is notified. URI of the diverted-to user not received.** The user A and user C are in network A. The user B is in network B and is provided with CFU Originating user receives notification that his communication has been diverted = Yes and ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" =No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and not informed of the diverted-to number and served user number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No
SIP Parameter	181 Being Forwarded History-Info: <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@networkA;cause=302?Privacy=history>;index=1.1
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFU is performed** ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) **Apply post test routine**
Comments	**Check:** A 181 Being Forwarded and a History-Info header is received at the interconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. **Check:** Is the cause parameter in the last entry is set to '302' **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfu_004
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	**Communication forwarding unconditional, originating user is notified. URI from the diverted-to user received.** The user A and user C are in network 1. The user B is in network N2 and is provided with CFU Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and informed of the diverted-to number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes
SIP Parameter	181 Being Forwarded History-Info: *<sip:userB@networkB>;index=1*, <sip: userC@networkA;cause=302>;index=1.1
Message flow	**SIP (Network A)** — **Interconnection Interface** — **SIP (Network B)** INVITE(Call-ID A-B) → **CFU is performed** ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) **Apply post test routine**
Comments	**Check:** A 181 Being Forwarded is received at the interconnection interface **Check:** A History-Info header is contained in the 181 with the URI of the diverted-to user. **Check:** Is the cause parameter in the last entry is set to '302'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfu_005
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	**Communication forwarding unconditional, diverted-to user does not receive the URI of the served user.** The user A and user C are in network A. The user B is in network B and is provided with CFU "Served user allows the presentation of his/her URI to the diverted-to user"= No. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user C is not informed of the forwarding number.
Configuration	**Subscription options:** • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE: Request line contains ';cause=302' History-Info header: <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@networkA;cause=302>;index=1.1
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFU is performed** ← INVITE(Call-ID B-C) **Apply post test routine**
Comments	**Check:** A History-Info header is received in the INVITE contains the URI of user B (served user) at the interconnection interface and a Privacy header is escaped set to 'history'. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '302'? **Check:** Is the cause parameter in the last entry is set to '302'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfu_006
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	**Communication forwarding unconditional, diverted-to user receives the URI of the served user.** The user A and user C are in network A. The user B is in network B and is provided with CFU "Served user allows the presentation of his/her URI to diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user C is informed of the forwarding number.
Configuration	**Subscription options:** • Served user allows the presentation of his/her URI to diverted-to user = Yes
SIP Parameter	INVITE: Request line contains ';cause=302' History-Info header: <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=302>;index=1.1
Message flow	
SIP (Network A)	**Interconnection Interface** / **SIP (Network B)**
	INVITE(Call-ID A-B) →
	CFU is performed
	← INVITE(Call-ID B-C)
	Apply post test routine
Comments	**Check:** A History-Info header is received in the INVITE contains the URI of user B (served user) at the interconnection interface. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '302'? **Check:** Is the cause parameter in the last entry is set to '302'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfu_007
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	**Communication forwarding unconditional, full notification.** The user A and user C are in network A. The user B is in network B and is provided with CFU Originating user receives notification that his communication has been diverted = Yes and ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, and "Served user allows the presentation of his/her URI to diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes • Served user allows the presentation of his/her URI to diverted-to user = Yes
SIP Parameter	INVITE: Request line contains ';cause=302' History-Info header: <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=302>;index=1.1 181 Being Forwarded History-Info header: <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=408>;index=1.1 200 OK INVITE History-Info header: <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=486>;index=1.1
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
	←	181 Being Forwarded(Call-ID B-A		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID C-B)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

Comments	**Check:** A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. **Check:** A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '302'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfu_008
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 25
Test purpose	**Communication forwarding unconditional, unsuccessful UDUB.** The user A and user C are in network A. The user B is in network B and is provided with CFU. Ensure that when user A calls user B, the call is forwarded unconditional to user C user C is user determined user busy.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
		486 Busy Here(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	486 Busy Here(Call-ID A-B)		
		ACK(Call-ID A-B)	→	

Comments	**Check:** The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction.

Test case number	SS_cfu_009
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 25
Test purpose	**Communication forwarding unconditional, unsuccessful NDUB.** The user A and user C are in network A. The user B is in network B. Ensure that when user A calls user B, the call is forwarded unconditional to user C and user C is network determined user busy.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
		486 Busy Here(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	486 Busy Here(Call-ID A-B)		
		ACK(Call-ID A-B)	→	

Comments	**Check:** The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction.

Test case number	SS_cfu_010
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 25 AND SE 30 AND [Network A] SE 9
Test purpose	**Communication forwarding unconditional, interaction with a not trusted network.** The user A and user C are in network A. The user B is in network B and is provided with CFU Originating user receives notification that his communication has been diverted = Yes ("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. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header

Message flow		
SIP (Network A)	**Interconnection Interface**	**SIP (Network B)**
	INVITE(Call-ID A-B) →	
	CFU is performed	
←	INVITE(Call-ID B-C)	
←	181 Being Forwarded(Call-ID B-A)	
	Apply post test routine	

Comments	**Check:** No History-Info header is received in the INVITE at the interconnection interface. **Check:** No History-Info header is received in the 181 Being Forwarded at the interconnection interface (if sent). Repeat this test in reverse direction.

Test case number	SS_cfu_011
Test case group	SIP-SIP/Service/CFU
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFU performed in Network B, Notification subscription options is set to presentation not allowed.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = no
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation not allowed Redirecting reason unconditional Generic notification call is diverting --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFU is performed** ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 183 Session Progress received at the interconnection interface? **Check:** Is an ACM encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator set to 'presentation not allowed'? **Check:** Is the Redirecting reason set to 'unconditional'? Repeat this test in reverse direction.

Test case number	SS_cfu_012
Test case group	SIP-SIP/Service/CFU
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFU performed in Network B, Notification subscription options is set to presentation allowed without redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed without redirection number Redirecting reason unconditional Generic notification call is diverting --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFU is performed** ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** 183 Session Progress is received at the interconnection interface. **Check:** Is an ACM encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator set to 'presentation allowed without redirection number'? **Check:** Is the Redirecting reason set to 'unconditional'? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_cfu_013</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CFU</td></tr>
<tr><td>Reference</td><td>6.5/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network B] SE 17 AND SE 47 AND SE 55</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. CFU performed in Network B, Notification subscription options is set to presentation allowed with redirection number.

The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number.
Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and informed of the diverted-to number.
The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.</td></tr>
<tr><td>Configuration</td><td>Subscription options:
• Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number</td></tr>
<tr><td>SIP Parameter</td><td>183 Session Progress
Content-Type: multipart/mixed;boundary=[any boundary name]

--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

ACM
Backward call indicator
Called party's status indicator
no indication
Redirection number
Address signal (Diverted-to user)
Call diversion information
Notification subscription options
presentation allowed with redirection number
Redirecting reason
unconditional
Generic notification
call is diverting

--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →
CFU is performed
← INVITE(Call-ID B-C, IAM)
← 183 Session Progress (Call-ID B-A, ACM)
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A
Check: 183 Session Progress is received at the interconnection interface.
Check: Is an ACM encapsulated in the 183?
Check: Is the Called party's status indicator set to 'no indication'?
Check: Is the Redirection number present?
Check: Is Notification subscription options indicator set to 'presentation allowed with redirection number'?
Check Is the Redirecting reason set to 'unconditional'?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_cfu_014
Test case group	SIP-SIP/Service/CFU
Reference	6.7/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. CFU performed in Network B, Restriction of the Redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Diverted-to user is subscribed to the COLR service in Permanent mode. Ensure that when user A calls user B, the call is forwarded unconditional to user C, a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	**Subscription options:** • Connected user subscribed to COLR, Permanent = yes
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** Redirection number restriction Presentation restricted --[any boundary name]--
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE(Call-ID A-B), IAM	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing (Call-ID C-B, ACM)	→	
	←	180 Ringing (Call-ID B-A)		
		200 OK INVITE (Call-ID C-B, ANM)	→	
	←	ACK (Call-ID B-C)		
	←	200 OK INVITE (Call-ID B-A)		
		ACK (Call-ID A-B)	→	
		Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 200 OK INVITE received at the interconnection interface? **Check:** Is an ANM encapsulated in the 200 OK? **Check:** Is the ISUP/BICC Redirection number restriction set to 'Presentation restricted'? Repeat this test in reverse direction.

Test case number	SS_cfu_015
Test case group	SIP-SIP/Service/CFU
Reference	6.7/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. CFU performed in Network B, No restriction of the Redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Diverted-to user is not subscribed to the COLR service. Ensure that when user A calls user B, the call is forwarded unconditional to user C, if a Redirection number restriction parameter is present it is set to 'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	**Subscription options:** • Connected user subscribed to COLR = no
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** Redirection number restriction Presentation allowed or Redirection number restriction not present --[any boundary name]--

Message flow

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B), IAM	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing (Call-ID C-B, ACM)	→	
	←	180 Ringing (Call-ID B-A)		
		200 OK INVITE (Call-ID C-B, ANM)	→	
	←	ACK (Call-ID B-C)		
	←	200 OK INVITE (Call-ID B-A)		
		ACK (Call-ID A-B)	→	
		Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 200 OK INVITE received at the interconnection interface? **Check:** Is an ANM encapsulated in the 200 OK? **Check:** Is the ISUP/BICC Redirection number restriction present set to 'Presentation allowed' or is the parameter absent? Repeat this test in reverse direction.

Test case number	SS_cfu_016
Test case group	SIP-SIP/Service/CFU
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFU performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.
Configuration	**Subscription options:** • Served user releases his/her number to diverted-to user = Release diverting number information
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Redirecting number Address presentation restricted indicator presentation allowed Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason unconditional --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFU is performed** ← INVITE(Call-ID B-C, IAM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a INVITE request received at the interconnection interface? **Check:** Is an IAM encapsulated in the INVITE? **Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? **Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? **Check:** Is the Redirection number present? **Check:** Is Redirection information present and the Redirecting reason is set to 'unconditional'? Repeat this test in reverse direction.

Test case number	SS_cfu_017
Test case group	SIP-SIP/Service/CFU
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFU performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Do not release diverting numberinformation
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Redirecting number Address presentation restricted indicator presentation restricted Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason unconditional --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFU is performed** ← INVITE(Call-ID B-C, IAM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** **Is a INVITE request received at the interconnection interface**? **Check:** Is an IAM encapsulated in the INVITE? **Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation restricted'? **Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation restricted'? **Check:** Is the Redirection number present? **Check:** Is Redirection information present and the Redirecting reason is set to 'unconditional'? Repeat this test in reverse direction.

7.1.5.6.2 Communication Forwarding Busy (CFB)

Test case number	SS_cfb_001
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 26
Test purpose	**Communication forwarding busy, basic rules.** The user A and user C are in Network A. The user B is in network B and is provided with CFB. Ensure that when user A calls user B, the call is forwarded busy to user C. In the active call state, ensure the property of speech.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE(Call-ID A-B)	→	
		CFB is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

Comments	**Check:** CDIV busy is successful. **Check:** In the active call state, ensure the property of speech. **Check:** Is the P-Asserted-Identity present set to the identity of the originating user? Repeat this test in reverse direction.

Test case number	SS_cfb_002
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	**Communication forwarding busy, no notification.** The user A and user C are in Network A. The user B is in network B and is provided with CFB, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B, the call is forwarded busy to user C, originating user is not notified.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = No
SIP Parameter	
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE(Call-ID A-B)	→	
		CFB is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		Apply post test routine		

Comments	**Check:** No notification regarding call forwarding in network B is received at the interconnection interface. Repeat this test in reverse direction.

Test case number	SS_cfb_003
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	**Communication forwarding busy, originating user is notified. URI from the served user not received.** The user A and user C are in network A. The user B is in network B and is provided with CFB Originating user receives notification that his communication has been diverted = Yes ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded busy to user C, user A is notified of call diversion and not informed of the diverted-to number and served user number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No
SIP Parameter	181 Being Forwarded <sip:userB@networkB?Privacy=history&Reason=SIP;cause=486>;index=1, <sip: userC@networkA;cause=486?*Privacy=history*>;index=1.1
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CFB is performed		
	←	INVITE(Call-ID B-C)		
	←	181 Being Forwarded(Call-ID B-A)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		Apply post test routine		

Comments	**Check:** A 181 Being Forwarded and a History-Info header is received at the interconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. **Check:** Is the cause parameter in the last entry set to '486'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfb_004
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	**Communication forwarding busy, originating user is notified. URI from the diverted-to user received.** The user A and user C are in network A. The user B is in network B and is provided with CFB Originating user receives notification that his communication has been diverted = Yes ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" =Yes. Ensure that when user A calls user B, the call is forwarded busy to user C, user A is notified of call diversion and informed of the diverted-to number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification =Yes
SIP Parameter	181 Being Forwarded <*sip:userB@networkB?*Reason=SIP; cause=486>*;index=1,* <sip: userC@networkA;cause=486>;index=1.1
Message flow	
SIP (Network A)	**Interconnection Interface** — **SIP (Network B)**
	INVITE(Call-ID A-B) →
	CFB is performed
	← INVITE(Call-ID B-C)
	← 181 Being Forwarded(Call-ID B-A)
	180 Ringing(Call-ID C-B) →
	← 180 Ringing(Call-ID B-A)
	Apply post test routine
Comments	**Check:** A 181 Being Forwarded is received at interconnection interface. **Check:** A History-Info header is contained in the 181 with the URI of the diverted-to user. **Check:** Is the cause parameter in the last entry set to '486'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_cfb_005</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CFB</td></tr>
<tr><td>Reference</td><td>4.5.2.6/[9]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>SE 26 AND SE 30</td></tr>
<tr><td>Test purpose</td><td>Communication forwarding busy, diverted-to user does not receive the URI of the served user.

The user A and user C are in network C. The user B is in network B and is provided with CFB "Served user allows the presentation of his/her URI to the diverted-to user" = No.
Ensure that when user A calls user B, the call is forwarded busy to user C, user C is not informed of the forwarding number.</td></tr>
<tr><td>Configuration</td><td>Subscription options:
• Served user allows the presentation of his/her URI to the diverted-to user = No</td></tr>
<tr><td>SIP Parameter</td><td>INVITE:
Request line contains ';cause=486'
History-Info header:
<sip:userB@networkB?Privacy=history&Reason=SIP;cause=486>;index=1,
<sip: userC@networkA;cause=486>;index=1.1</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →
CFB is performed
← INVITE(Call-ID B-C)
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Check: A History-Info header is received in the INVITE contains the URI of user B (served user) at the interconnection interface and a Privacy header is escaped set to 'history'.
Check: Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '486'?
Check: Is the cause parameter in the last entry set to '486'?
NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_cfb_006
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	**Communication forwarding busy, diverted-to user receives the URI of the served user.** The user A and user C are in network C. The user B is in network B and is provided with CFB "Served user allows the presentation of his/her URI to the diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded busy to user C, user C is informed of the forwarding number.
Configuration	**Subscription options:** • Served user allows the presentation of his/her URI to the diverted-to user = Yes
SIP Parameter	INVITE: Request line contains ';cause=486' History-Info header: <sip:userB@networkB?Reason=SIP;cause=486>;index=1, <sip: userC@networkA;cause=486>;index=1.1
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFB is performed** ← INVITE(Call-ID B-C) **Apply post test routine**
Comments	**Check:** A History-Info header is received in the INVITE contains the URI of user B (served user) at the interconnection interface. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '486'? **Check:** Is the cause parameter in the last entry set to '486'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfb_007
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	**Communication forwarding busy, full notification.** The user A and user C are in network A. The user B is in network B and is provided with CFB Originating user receives notification that his communication has been diverted = Yes ("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. Ensure that when user A calls user B, the call is forwarded busy to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • 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
SIP Parameter	INVITE: Request line contains ';cause=486' History-Info header: <sip:userB@networkB&Reason=SIP;cause=486>;index=1, <sip: userC@networkA;cause=486>;index=1.1 181 Being Forwarded History-Info header: <sip:userB@networkB&Reason=SIP;cause=486>;index=1, <sip: userC@networkA;cause=486>;index=1.1 200 OK INVITE History-Info header: <sip:userB@networkB&Reason=SIP;cause=486>;index=1, <sip: userC@networkA;cause=486>;index=1.1
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFB is performed** ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ← ACK(Call-ID C-B) ← 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → **Communication** **Apply post test routine**
Comments	**Check:** A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. **Check:** A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '486'? **Check:** Is the cause parameter in the last entry set to '486'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfb_008
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 26
Test purpose	**Communication forwarding busy, unsuccessful UDUB.** The user A and user C are in network A. The user B is in network B and is provided with CFB. Ensure that when user A calls user B, the call is forwarded busy to user C and user C is user determined user busy.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFB is performed** ← INVITE(Call-ID B-C) 486 Busy Here(Call-ID C-B) → ← ACK(Call-ID B-C) ← 486 Busy Here(Call-ID A-B) ACK(Call-ID A-B) →
Comments	**Check:** The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction.

Test case number	SS_cfb_009
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 26
Test purpose	**Communication forwarding busy, unsuccessful NDUB.** The user A and user C are in network A. The user B is in network B and is provided with CFB. Ensure that when user A calls user B, the call is forwarded busy to user C and user C is network determined user busy.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFB is performed** ← INVITE(Call-ID B-C) 486 Busy Here(Call-ID C-B) → ← ACK(Call-ID B-C) ← 486 Busy Here(Call-ID A-B) ACK(Call-ID A-B) →
Comments	**Check:** A 181 Being Forwarded is received at network 1 originating access. **Check:** The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction.

Test case number	SS_cfb_010
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 26 AND SE 30 AND [Network A] SE 9
Test purpose	**Communication forwarding busy, interaction with a not trusted network.** The user A and user C are in network A. The user B is in network B and is provided with CFB Originating user receives notification that his communication has been diverted = Yes ("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. Ensure that when user A calls user B, the call is forwarded busy to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header

Message flow

SIP (Network A)	Interconnection Interface		SIP (Network B)
	INVITE(Call-ID A-B)	→	
	CFB is performed		
←	INVITE(Call-ID B-C)		
←	181 Being Forwarded(Call-ID B-A)		
	Apply post test routine		

Comments	**Check:** No History-Info header is received in the INVITE at the interconnection interface. **Check:** No History-Info header is received in the 181 Being Forwarded at the interconnection interface (if sent). Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_cfb_011</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CFB</td></tr>
<tr><td>Reference</td><td>6.5/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network B] SE 17 AND SE 47 AND SE 55</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. CFB performed in Network B, Notification subscription options is set to presentation not allowed.

The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number.
Ensure that when user A calls user B, the call is forwarded on busy user to user C, user A is not notified about call diversion.
The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.</td></tr>
<tr><td>Configuration</td><td>Subscription options:
• Calling user receives notification that his call has been diverted (forwarded or deflected) = no</td></tr>
<tr><td>SIP Parameter</td><td>183 Session Progress
Content-Type: multipart/mixed;boundary=[any boundary name]

--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

ACM
Backward call indicator
Called party's status indicator
no indication
Redirection number
Address signal (Diverted-to user)
Call diversion information
Notification subscription options
presentation not allowed
Redirecting reason
User Busy
Generic notification
call is diverting

--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →
CFB is performed
← INVITE(Call-ID B-C, IAM)
← 183 Session Progress (Call-ID B-A, ACM)
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A
Check: Is a 183 Session Progress received at the interconnection interface?
Check: Is an ACM encapsulated in the 183?
Check: Is the Called party's status indicator set to 'no indication'?
Check: Is the Redirection number present?
Check: Is Notification subscription options indicator set to 'presentation not allowed'?
Check: Is the Redirecting reason set to User Busy'?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_cfb_012
Test case group	SIP-SIP/Service/CFB
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFB performed in Network B, Notification subscription options is set to presentation allowed without redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on busy user to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed without redirection number Redirecting reason User Busy Generic notification call is diverting --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFB is performed** ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** 183 Session Progress is received at the interconnection interface. **Check:** Is an ACM encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator is set to 'presentation allowed without redirection number'? **Check:** Is the Redirecting reason set to 'User Busy'? Repeat this test in reverse direction.

Test case number	SS_cfb_013
Test case group	SIP-SIP/Service/CFB
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFB performed in Network B, Notification subscription options is set to presentation allowed with redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number. Ensure that when user A calls user B, the call is forwarded on busy user to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** Backward call indicator Called party's status indicator no indication Redirection number (*Diverted-to user*) Address signal Call diversion information Notification subscription options presentation allowed with redirection number Redirecting reason User Busy Generic notification call is diverting --[any boundary name]--
Message flow	

SIP (Network A)	Interconnection Interface		SIP (Network B)
	INVITE(Call-ID A-B)	→	
	CFB is performed		
←	INVITE(Call-ID B-C, IAM)		
←	183 Session Progress (Call-ID B-A, ACM)		
	Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** 183 Session Progress is received at the interconnection interface. **Check:** Is an ACM encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator is set to 'presentation allowed with redirection number'? **Check:** Is the Redirecting reason set to 'User Busy'? Repeat this test in reverse direction.

Test case number	SS_cfb_014
Test case group	SIP-SIP/Service/CFB
Reference	6.7/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. CFB performed in Network B, Restriction of the Redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Diverted-to user is subscribed to the COLR service in Permanent mode. Ensure that when user A calls user B, the call is forwarded on busy user to user C, a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	**Subscription options:** • Connected user subscribed to COLR, Permanent = yes
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** Redirection number restriction Presentation restricted --[any boundary name]--

Message flow			
SIP (Network A)	**Interconnection Interface**		**SIP (Network B)**
	INVITE(Call-ID A-B), IAM	→	
	CFB is performed		
←	INVITE(Call-ID B-C)		
	180 Ringing (Call-ID C-B, ACM)	→	
←	180 Ringing (Call-ID B-A)		
	200 OK INVITE (Call-ID C-B, ANM)	→	
←	ACK (Call-ID B-C)		
←	200 OK INVITE (Call-ID B-A)		
	ACK (Call-ID A-B)	→	
	Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 200 OK INVITE received at the interconnection interface? **Check:** Is an ANM encapsulated in the 200 OK? **Check:** Is the ISUP/BICC Redirection number restriction set to 'Presentation restricted'? Repeat this test in reverse direction.

Test case number	SS_cfb_015
Test case group	SIP-SIP/Service/CFB
Reference	6.7/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. CFB performed in Network B, No restriction of the Redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Diverted-to user is not subscribed to the COLR service. Ensure that when user A calls user B, the call is forwarded on busy user to user C, if a Redirection number restriction parameter is present it is set to 'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	**Subscription options:** • Connected user subscribed to COLR = no
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** Redirection number restriction Presentation allowed or Redirection number restriction not present --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B), IAM → **CFB is performed** ← INVITE(Call-ID B-C) 180 Ringing (Call-ID C-B, ACM) → ← 180 Ringing (Call-ID B-A) 200 OK INVITE (Call-ID C-B, ANM) → ← ACK (Call-ID B-C) ← 200 OK INVITE (Call-ID B-A) ACK (Call-ID A-B) → **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 200 OK INVITE received at the interconnection interface? **Check:** Is an ANM encapsulated in the 200 OK? **Check:** Is the ISUP/BICC Redirection number restriction present set to 'Presentation allowed' or is the parameter absent? Repeat this test in reverse direction.

Test case number	SS_cfb_016
Test case group	SIP-SIP/Service/CFB
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFB performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded on busy user to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.
Configuration	**Subscription options:** • Served user releases his/her number to diverted-to user = Release diverting number information
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Redirecting number Address presentation restricted indicator presentation allowed Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason User Busy --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFB is performed** ← INVITE(Call-ID B-C, IAM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** **Is a INVITE request received at the interconnection interface**? **Check:** Is an IAM encapsulated in the INVITE? **Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? **Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? **Check:** Is the Redirection number present? **Check:** Is Redirection information present and the Redirecting reason is set to 'User Busy'? Repeat this test in reverse direction.

Test case number	SS_cfb_017
Test case group	SIP-SIP/Service/CFB
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFB performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded on busy user to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Do not release diverting numberinformation
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Redirecting number Address presentation restricted indicator presentation restricted Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason User Busy --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFB is performed** ← INVITE(Call-ID B-C, IAM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** **Is a INVITE request received at the interconnection interface**? **Check:** Is an IAM encapsulated in the INVITE? **Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation restricted'? **Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation restricted'? **Check:** Is the Redirection number present? **Check:** Is Redirection information present and the Redirecting reason is set to 'User Busy'? Repeat this test in reverse direction.

7.1.5.6.3 Communication Forwarding No Reply (CFNR)

Test case number	SS_cfnr_001		
Test case group	SIP-SIP/Service/CFNR		
Reference	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 27		
Test purpose	**Communication forwarding no reply, basic rules.** The user A and user C are in Network A. The user B is in network B and is provided with CFNR. Ensure that when user A calls user B, the call is forwarded no reply to user C. In the active call state, ensure the property of speech.		
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)	**Interconnection Interface**		**SIP (Network B)**
	INVITE(Call-ID A-B)	→	
←	180 Ringing(Call-ID B-A)		
	CFB is performed		
←	INVITE(Call-ID B-C)		
	180 Ringing(Call-ID C-B)	→	
←	180 Ringing(Call-ID B-A)		
	200 OK INVITE(Call-ID C-B)	→	
←	ACK(Call-ID B-C)		
←	200 OK INVITE(Call-ID B-A)		
	ACK(Call-ID A-B)	→	
	Communication		
	Apply post test routine		
Comments	**Check:** CDIV no reply is successful. **Check:** In the active call state, ensure the property of speech. **Check:** Is the P-Asserted-Identity present set to the identity of the originating user? Repeat this test in reverse direction.		

Test case number	SS_cfnr_002		
Test case group	SIP-SIP/Service/CFNR		
Reference	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 27 AND SE 30		
Test purpose	**Communication forwarding no reply, no notification.** The user A and user C are in Network A. The user B is in network B and is provided with CFNR, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B, the call is forwarded no reply to user C, originating user is not notified.		
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = No		
SIP Parameter			
Message flow			
SIP (Network A)	**Interconnection Interface**		**SIP (Network B)**
	INVITE(Call-ID A-B)	→	
←	180 Ringing(Call-ID B-A)		
	CFB is performed		
←	INVITE(Call-ID B-C)		
	180 Ringing(Call-ID C-B)	→	
←	180 Ringing(Call-ID B-A)		
	Apply post test routine		
Comments	**Check:** No notification regarding call forwarding in network B is received at the interconnection interface. Repeat this test in reverse direction.		

Test case number	SS_cfnr_003
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 27 AND SE 30
Test purpose	**Communication forwarding no reply, originating user is notified. URI from the served user not received.** The user A and user C are in network A. The user B is in network B and is provided with CFNR Originating user receives notification that his communication has been diverted = Yes ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded no reply to user C, user A is notified of call diversion and not informed of the diverted-to number and served user number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No
SIP Parameter	181 Being Forwarded <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@networkA;cause=408?Privacy=history>;index=1.1
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → ← 180 Ringing(Call-ID B-A) **CFB is performed** ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) **Apply post test routine**
Comments	**Check:** A 181 Being Forwarded and a History-Info header is received at the interconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. **Check:** Is the cause parameter in the last entry is set to '408'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfnr_004
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 27 AND SE 30
Test purpose	**Communication forwarding no reply, originating user is notified. URI from the diverted-to user received.** The user A and user C are in network A. The user B is in network B and is provided with CFNR Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" =Yes. Ensure that when user A calls user B, the call is forwarded no reply to user C, user A is notified of call diversion and informed of the diverted-to number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification =Yes
SIP Parameter	181 Being Forwarded <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=408>;index=1.1
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
	←	180 Ringing(Call-ID B-A)		
		CFB is performed		
	←	INVITE(Call-ID B-C)		
	←	181 Being Forwarded(Call-ID B-A)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		Apply post test routine		

Comments	**Check:** A 181 Being Forwarded is received at the interconnection interface. **Check:** A History-Info header is contained in the 181 with the URI of the diverted-to user. **Check:** Is the cause parameter in the last entry is set to '408'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfnr_005
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 27 AND SE 30
Test purpose	**Communication forwarding no reply, diverted-to user does not receive the URI of the served user.** The user A and user C are in network A. The user B is in network B and is provided with "Served user allows the presentation of his/her URI to the diverted-to user" = No. Ensure that when user A calls user B, the call is forwarded no reply to user C, user C is not informed of the forwarding number.
Configuration	**Subscription options:** • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE Request line contains ';cause=408' History-Info header: <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@network1;cause=408>;index=1.1
Message flow	
SIP (Network A)	**Interconnection Interface** — **SIP (Network B)** INVITE(Call-ID A-B) → ← 180 Ringing(Call-ID B-A) **CFB is performed** ← INVITE(Call-ID B-C) **Apply post test routine**
Comments	**Check:** A History-Info header is received in the INVITE contains the URI of user B (served user) at the interconnection interface and a Privacy header is escaped set to 'history'. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '408'? **Check:** Is the cause parameter in the last entry is set to '408'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfnr_006
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 27 AND SE 30
Test purpose	**Communication forwarding no reply, diverted-to user receives the URI of the diverted-to user.** The user A and user C are in network A. The user B is in network B and is provided with "Served user allows the presentation of his/her URI to the diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded no reply to user C, user C is informed of the forwarding number.
Configuration	**Subscription options:** • Served user allows the presentation of his/her URI to the diverted-to user = Yes
SIP Parameter	INVITE Request line contains ';cause=408' History-Info header: <sip:userB@networkB>;index=1, <sip: userC@network1;cause=408>;index=1.1
Message flow	
SIP (Network A)	**Interconnection Interface** — **SIP (Network B)** INVITE(Call-ID A-B) → ← 180 Ringing(Call-ID B-A) **CFB is performed** ← INVITE(Call-ID B-C) **Apply post test routine**
Comments	

Test case number	SS_cfnr_007
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 27 AND SE 30
Test purpose	**Communication forwarding no reply, full notification.** The user A and user C are in network A. The user B is in network B and is provided with CFNR Originating user receives notification that his communication has been diverted = Yes, ("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. Ensure that when user A calls user B, the call is forwarded no reply to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • 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
SIP Parameter	INVITE: Request line contains ';cause=408' History-Info header: <sip:userB@networkB&Reason=SIP;cause=408>;index=1, <sip: userC@networkA;cause=486>;index=1.1 181 Being Forwarded History-Info header: <sip:userB@network>;index=1, <sip: userC@networkA;cause=408>;index=1.1 200 OK INVITE History-Info header: <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=408>;index=1.1
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
	←	180 Ringing(Call-ID B-A)		
		CFB is performed		
	←	INVITE(Call-ID B-C)		
	←	181 Being Forwarded(Call-ID B-A		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID C-B)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Apply post test routine		

Comments	**Check:** A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. **Check:** A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '408'? **Check:** Is the cause parameter in the last entry is set to '408'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfnr_008
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 27
Test purpose	**Communication forwarding no reply, unsuccessful UDUB.** The user A and user C are in network A. The user B is in network B and is provided with CFNR. Ensure that when user A calls user B, the call is forwarded no reply to user C and user C is user determined user busy.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE(Call-ID A-B)	→	
	←	180 Ringing(Call-ID B-A)		
		CFB is performed		
	←	INVITE(Call-ID B-C)		
		486 Busy Here(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	486 Busy Here(Call-ID A-B)		
		ACK(Call-ID A-B)	→	

Comments	**Check:** The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction.

Test case number	SS_cfnr_009
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 27
Test purpose	**Communication forwarding no reply, unsuccessful NDUB.** The user A and user C are in network A. The user B is in network B and is provided with CFNR. Ensure that when user A calls user B, the call is forwarded no reply to user C and user C is network determined user busy.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE(Call-ID A-B)	→	
	←	180 Ringing(Call-ID B-A)		
		CFB is performed		
	←	INVITE(Call-ID B-C)		
		486 Busy Here(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	486 Busy Here(Call-ID A-B)		
		ACK(Call-ID A-B)	→	

Comments	**Check:** The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction.

Test case number	SS_cfnr_010
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 27 AND SE 30 AND [Network A] is SE 9
Test purpose	**Communication forwarding no reply, interaction with a not trusted network.** The user A and user C are in network A. The user B is in network B and is provided with CFNR Originating user receives notification that his communication has been diverted = Yes ("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. Ensure that when user A calls user B, the call is forwarded no reply to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → ← 180 Ringing(Call-ID B-A) **CFB is performed** ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) **Apply post test routine**
Comments	**Check:** No History-Info header is received in the INVITE at the interconnection interface. **Check:** No History-Info header is received in the 181 Being Forwarded at the interconnection interface (if sent). Repeat this test in reverse direction.

Test case number	SS_cfnr_011
Test case group	SIP-SIP/Service/CFNR
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFNR performed in Network B, Notification subscription options is set to presentation not allowed.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on no reply to user C, user A is not notified about call diversion. The notification information is present in the encapsulated CPG contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = no
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Event indicator Alerting or Progress Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation not allowed Redirecting reason No reply Generic notification call is diverting --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A, ACM) **CFNR is performed** ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, CPG) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 183 Session Progress received at the interconnection interface? **Check:** Is an CPG encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator set to 'presentation not allowed'? **Check:** Is the Redirecting reason set to 'No reply'? Repeat this test in reverse direction.

Test case number	SS_cfnr_012
Test case group	SIP-SIP/Service/CFNR
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFNR performed in Network B, Notification subscription options is set to presentation allowed without redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on no reply to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated CPG contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Event indicator Alerting or Progress Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed without redirection number Redirecting reason No reply Generic notification call is diverting --[any boundary name]--
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
	←	180 Ringing (Call-ID B-A, ACM)		
		CFNR is performed		
	←	INVITE(Call-ID B-C, IAM)		
	←	183 Session Progress (Call-ID B-A, ACM)		
		Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** 183 Session Progress is received at the interconnection interface. **Check:** Is an CPG encapsulated in the 183? **Check:** is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator is set to 'presentation allowed without redirection number'? **Check:** Is the Redirecting reason set to 'No reply'? Repeat this test in reverse direction.

Test case number	SS_cfnr_013
Test case group	SIP-SIP/Service/CFNR
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFNR performed in Network B, Notification subscription options is set to presentation allowed with redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number. Ensure that when user A calls user B, the call is forwarded on no reply to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated CPG contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Event indicator Alerting or Progress Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed with redirection number Redirecting reason No reply Generic notification call is diverting --[any boundary name]--
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE(Call-ID A-B)	→	
	←	180 Ringing (Call-ID B-A, ACM)		
		CFNR is performed		
	←	INVITE(Call-ID B-C, IAM)		
	←	183 Session Progress (Call-ID B-A, ACM)		
		Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** 183 Session Progress is received at the interconnection interface. **Check:** Is an CPG encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator is set to 'presentation allowed with redirection number'? **Check:** Is the Redirecting reason set to 'No reply'? Repeat this test in reverse direction.

Test case number	SS_cfnr_014
Test case group	SIP-SIP/Service/CFNR
Reference	6.7/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. CFNR performed in Network B, Restriction of the Redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Diverted-to user is subscribed to the COLR service in Permanent mode. Ensure that when user A calls user B, the call is forwarded on no reply to user C, a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	**Subscription options:** • Connected user subscribed to COLR, Permanent = yes
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** Redirection number restriction Presentation restricted --[any boundary name]--
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B), IAM	→	
	←	180 Ringing (Call-ID B-A, ACM)		
		CFNR is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing (Call-ID C-B, ACM)	→	
	←	180 Ringing (Call-ID B-A)		
		200 OK INVITE (Call-ID C-B, ANM)	→	
	←	ACK (Call-ID B-C)		
	←	200 OK INVITE (Call-ID B-A)		
		ACK (Call-ID A-B)	→	
		Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 200 OK INVITE received at the interconnection interface? **Check:** Is an ANM encapsulated in the 200 OK? **Check:** Is the ISUP/BICC Redirection number restriction set to 'Presentation restricted'? Repeat this test in reverse direction.

Test case number	SS_cfnr_015
Test case group	SIP-SIP/Service/CFNR
Reference	6.7/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. CFNR performed in Network B, No restriction of the Redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Diverted-to user is not subscribed to the COLR service. Ensure that when user A calls user B, the call is forwarded on no reply to user C, if a Redirection number restriction parameter is present it is set to 'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	**Subscription options:** • Connected user subscribed to COLR = no
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** Redirection number restriction Presentation allowed or Redirection number restriction not present --[any boundary name]--
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE(Call-ID A-B), IAM	→	
	←	180 Ringing (Call-ID B-A)		
		CFNR is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing (Call-ID C-B, ACM)	→	
	←	180 Ringing (Call-ID B-A)		
		200 OK INVITE (Call-ID C-B, ANM)	→	
	←	ACK (Call-ID B-C)		
	←	200 OK INVITE (Call-ID B-A)		
		ACK (Call-ID A-B)	→	
		Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 200 OK INVITE received at the interconnection interface? **Check:** Is an ANM encapsulated in the 200 OK? **Check:** Is the ISUP/BICC Redirection number restriction present set to 'Presentation allowed' or is the parameter absent? Repeat this test in reverse direction.

Test case number	SS_cfnr_016
Test case group	SIP-SIP/Service/CFNR
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFNR performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded on no reply to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.
Configuration	**Subscription options:** • Served user releases his/her number to diverted-to user = Release diverting number information
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Redirecting number Address presentation restricted indicator presentation allowed Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason No reply --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A, ACM) **CFNR is performed** ← INVITE(Call-ID B-C, IAM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** **Is a INVITE request received at the interconnection interface**? **Check:** Is an IAM encapsulated in the INVITE? **Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? **Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? **Check:** Is the Redirection number present? **Check:** Is Redirection information present and the Redirecting reason is set to 'No reply'? Repeat this test in reverse direction.

Test case number	SS_cfnr_017
Test case group	SIP-SIP/Service/CFNR
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFNR performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded on no reply to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Do not release diverting numberinformation
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Redirecting number Address presentation restricted indicator presentation restricted Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason No reply --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A, ACM) **CFNR is performed** ← INVITE(Call-ID B-C, IAM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** **Is a INVITE request received at the interconnection interface**? **Check:** Is an IAM encapsulated in the INVITE? **Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation restricted'? **Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation restricted'? **Check:** Is the Redirection number present? **Check:** Is Redirection information present and the Redirecting reason is set to 'No reply'? Repeat this test in reverse direction.

7.1.5.6.4 Communication Forwarding Not Logged in (CFNL)

Test case number	SS_cfnl_001
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 28
Test purpose	**Communication forwarding not logged in, basic rules.** The user A and user C are in Network A. The user B is in network B and is provided with CFNL. Ensure that when user A calls user B, the call is forwarded not logged in to user C. In the active call state, ensure the property of speech.
Configuration	
SIP Parameter	
Message flow	
SIP (Network A)	**Interconnection Interface** / **SIP (Network B)**
	INVITE(Call-ID A-B) →
	CFNL is performed
	← INVITE(Call-ID B-C)
	180 Ringing(Call-ID C-B) →
	← 180 Ringing(Call-ID B-A)
	200 OK INVITE(Call-ID C-B) →
	← ACK(Call-ID B-C)
	← 200 OK INVITE(Call-ID B-A)
	ACK(Call-ID A-B) →
	Communication
	Apply post test routine
Comments	**Check:** The CDIV not logged in is successful. **Check:** In the active call state, ensure the property of speech. **Check:** Is the P-Asserted-Identity present set to the identity of the originating user? Repeat this test in reverse direction.

Test case number	SS_cfnl_002
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	**Communication forwarding not logged in, no notification.** The user A and user C are in Network A. The user B is in network B and is provided with CFNL, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, originating user is not notified.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = No
SIP Parameter	
Message flow	
SIP (Network A)	**Interconnection Interface** / **SIP (Network B)**
	INVITE(Call-ID A-B) →
	CFNL is performed
	← INVITE(Call-ID B-C)
	180 Ringing(Call-ID C-B) →
	← 180 Ringing(Call-ID B-A)
	Apply post test routine
Comments	**Check:** No notification regarding call forwarding in network B is received at interconnection interface. Repeat this test in reverse direction.

Test case number	SS_cfnl_003
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	**Communication forwarding not logged in, originating user is notified. URI of the diverted-to user not received.** The user A and user C are in network A. The user B is in network B and is provided with CFNL Originating user receives notification that his communication has been diverted = Yes and ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and not informed of the diverted-to number and the served user number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No
SIP Parameter	181 Being Forwarded *<sip:userB@networkB?Privacy=history>;index=1,* *<sip: userC@networkA;cause=404?Privacy=history>;index=1.1*
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFNL is performed** ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) **Apply post test routine**
Comments	**Check:** A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. **Check:** is the cause parameter in the last entry is set to '404'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfnl_004
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	**Communication forwarding not logged in, originating user is notified. URI from the diverted-to user received.** The user A and user C are in network A. The user B is in network B and is provided with CFNL Originating user receives notification that his communication has been diverted = Yes and ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and informed of the diverted-to number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes
SIP Parameter	181 Being Forwarded *<sip:userB@networkB>;index=1,* <sip: userC@networkA;cause=404>;index=1.1
Message flow	
SIP (Network A)	**Interconnection Interface** — **SIP (Network B)**
	INVITE(Call-ID A-B) →
	CFNL is performed
	← INVITE(Call-ID B-C)
	← 181 Being Forwarded(Call-ID B-A)
	180 Ringing(Call-ID C-B) →
	← 180 Ringing(Call-ID B-A)
	Apply post test routine
Comments	**Check:** A 181 Being Forwarded is received at interconnection interface. **Check:** A History-Info header is contained in the 181 with the URI of the served user and the URI of the diverted-to user. **Check:** Is the cause parameter in the last entry is set to '404'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfnl_005
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	**Communication forwarding not logged in, diverted-to user does not receive the URI of the diverted-to user.** The user A and user C are in network A. The user B is in network B and is provided with CFNL "Served user allows the presentation of his/her URI to diverted-to user" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user C is not informed of the forwarding number.
Configuration	**Subscription options:** • Served user allows the presentation of his/her URI to diverted-to user = No
SIP Parameter	INVITE Request line contains ';cause=404' History-Info header: <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@network1;cause=404>;index=1.1
Message flow	
SIP (Network A)	**Interconnection Interface** / **SIP (Network B)** INVITE(Call-ID A-B) → **CFNL is performed** ← INVITE(Call-ID B-C) **Apply post test routine**
Comments	**Check:** A History-Info header is received in the INVITE contains the URI of user B (served user) at the interconnection interface and a Privacy header is escaped set to 'history'. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '404'? **Check:** Is the cause parameter in the last entry is set to '404'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfnl_006
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	**Communication forwarding not logged in, diverted-to user receives the URI of the served user.** The user A and user C are in network A. The user B is in network B and is provided with CFNL "Served user allows the presentation of his/her URI to diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user C is informed of the forwarding number.
Configuration	**Subscription options:** • Served user allows the presentation of his/her URI to diverted-to user = Yes
SIP Parameter	INVITE Request line contains ';cause=404' History-Info header: *<sip:userB@networkB>;index=1,* <sip: userC@networkA;cause=404>;index=1.1
Message flow	

SIP (Network A)	**Interconnection Interface**		**SIP (Network B)**
	INVITE(Call-ID A-B)	→	
	CFNL is performed		
←	INVITE(Call-ID B-C)		
	Apply post test routine		

Comments	**Check:** A History-Info header is received in the INVITE contains the URI of user B (served user) at the interconnection interface. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '404'? **Check:** Is the cause parameter in the last entry is set to '404'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfnl_007
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	**Communication forwarding not logged in, full notification.** The user A and user C are in network A. The user B is in network B and is provided with CFNL Originating user receives notification that his communication has been diverted = Yes, ("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. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • 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
SIP Parameter	INVITE: Request line contains ';cause=404' History-Info header: <sip:userB@networkB&Reason=SIP;cause=404>;index=1, <sip: userC@networkA;cause=404>;index=1.1 181 Being Forwarded History-Info header: <sip:userB@network>;index=1, <sip: userC@networkA;cause=404>;index=1.1 200 OK INVITE History-Info header: <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=404>;index=1.1

Message flow

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	➔	
		CFNL is performed		
	←	INVITE(Call-ID B-C)		
	←	181 Being Forwarded(Call-ID B-A		
		180 Ringing(Call-ID C-B)	➔	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	➔	
	←	ACK(Call-ID C-B)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	➔	
		Apply post test routine		

Comments	**Check:** A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. **Check:** A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '404'? **Check:** Is the cause parameter in the last entry is set to '404'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cfnl_008
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 28
Test purpose	**Communication forwarding not logged in, unsuccessful UDUB.** The user A and user C are in network A. The user B is in network B and is provided with CFNL. Ensure that when user A calls user B, the call is forwarded not logged in to user C and user C is user determined user busy.
Configuration	
SIP Parameter	
Message flow	**SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFNL is performed** 486 Busy Here(Call-ID C-B) → ← ACK(Call-ID B-C) ← 486 Busy Here(Call-ID A-B) ACK(Call-ID A-B) →
Comments	**Check:** The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction.

Test case number	SS_cfnl_009
Test case group	4.5.2.6/[9]
Reference	ES 183 004
SELECTION EXPRESSION	SE 28
Test purpose	**Communication forwarding not logged in, unsuccessful NDUB.** The user A and user C are in network A. The user B is in network B and is provided with CFNL. Ensure that when user A calls user B, the call is forwarded not logged in to user C and user C is busy.
Configuration	
SIP Parameter	
Message flow	**SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFNL is performed** 486 Busy Here(Call-ID C-B) → ← ACK(Call-ID B-C) ← 486 Busy Here(Call-ID A-B) ACK(Call-ID A-B) →
Comments	**Check:** The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction.

Test case number	SS_cfnl_010
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 28 AND SE 30 AND [Network A] SE 9
Test purpose	**Communication forwarding not logged in, interaction with a not trusted network.** The user A and user C are in network A. The user B is in network B and is provided with CFNL Originating user receives notification that his communication has been diverted = Yes ("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. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFNL is performed** ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) **Apply post test routine**
Comments	**Check:** No History-Info header is received in the INVITE at the interconnection interface. **Check:** No History-Info header is received in the 181 Being Forwarded at the interconnection interface (if sent). Repeat this test in reverse direction.

Test case number	SS_cfnl_011
Test case group	SIP-SIP/Service/CFNL
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFNL performed in Network B, Notification subscription options is set to presentation not allowed.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = no
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation not allowed Redirecting reason Mobile subscriber not reachable Generic notification call is diverting --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFNL is performed** ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 183 Session Progress received at the interconnection interface? **Check:** Is an ACM encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator set to 'presentation not allowed'? **Check:** Is the Redirecting reason set to 'Mobile subscriber not reachable'? Repeat this test in reverse direction.

Test case number	SS_cfnl_012
Test case group	SIP-SIP/Service/CFNL
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFNL performed in Network B, Notification subscription options is set to presentation allowed without redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed without redirection number Redirecting reason Mobile subscriber not reachable Generic notification call is diverting --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFNL is performed** ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** 183 Session Progress is received at the interconnection interface. **Check:** Is an ACM encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator is set to 'presentation allowed without redirection number'? **Check:** Is the Redirecting reason set to 'Mobile subscriber not reachable'? Repeat this test in reverse direction.

Test case number	SS_cfnl_013
Test case group	SIP-SIP/Service/CFNL
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFNL performed in Network B, Notification subscription options is set to presentation allowed with redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number. Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed with redirection number Redirecting reason Mobile subscriber not reachable Generic notification call is diverting --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFNL is performed** ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** 183 Session Progress is received at the interconnection interface **Check:** Is an ACM encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator is set to 'presentation allowed with redirection number'? **Check:** Is the Redirecting reason set to 'Mobile subscriber not reachable'? Repeat this test in reverse direction.

Test case number	SS_cfnl_014
Test case group	SIP-SIP/Service/CFNL
Reference	6.7/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. CFNL performed in Network B, Restriction of the Redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Diverted-to user is subscribed to the COLR service in Permanent mode. Ensure that when user A calls user B, the call is forwarded not logged in to user C, a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	**Subscription options:** • Connected user subscribed to COLR, Permanent = yes
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** Redirection number restriction Presentation restricted --[any boundary name]--

Message flow

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B), IAM	→	
		CFNL is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing (Call-ID C-B, ACM)	→	
	←	180 Ringing (Call-ID B-A)		
		200 OK INVITE (Call-ID C-B, ANM)	→	
	←	ACK (Call-ID B-C)		
	←	200 OK INVITE (Call-ID B-A)		
		ACK (Call-ID A-B)	→	
		Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 200 OK INVITE received at the interconnection interface **Check:** Is an ANM encapsulated in the 200 OK? **Check:** Is the ISUP/BICC Redirection number restriction set to 'Presentation restricted'? Repeat this test in reverse direction.

Test case number	SS_cfnl_015
Test case group	SIP-SIP/Service/CFNL
Reference	6.7/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. CFNL performed in Network B, No restriction of the Redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Diverted-to user is not subscribed to the COLR service. Ensure that when user A calls user B, the call is forwarded not logged in to user C, if a Redirection number restriction parameter is present it is set to 'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	**Subscription options:** • Connected user subscribed to COLR = no
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** Redirection number restriction Presentation allowed or Redirection number restriction not present --[any boundary name]--
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B), IAM	→	
		CFNL is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing (Call-ID C-B, ACM)	→	
	←	180 Ringing (Call-ID B-A)		
		200 OK INVITE (Call-ID C-B, ANM)	→	
	←	ACK (Call-ID B-C)		
	←	200 OK INVITE (Call-ID B-A)		
		ACK (Call-ID A-B)	→	
		Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 200 OK INVITE received at the interconnection interface? **Check:** Is an ANM encapsulated in the 200 OK? **Check:** Is the ISUP/BICC Redirection number restriction present set to 'Presentation allowed' or is the parameter absent? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_cfnl_016</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CFNL</td></tr>
<tr><td>Reference</td><td>7.1/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network B] SE 17 AND SE 47 AND SE 55</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. CFNL performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.

The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Served user releases his/her number to diverted-to user = Release diverting number information.
Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user C is notified of call diversion and informed of the diverting number.
The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.</td></tr>
<tr><td>Configuration</td><td>Subscription options:
• Served user releases his/her number to diverted-to user = Release diverting number information</td></tr>
<tr><td>SIP Parameter</td><td>INVITE
Content-Type: multipart/mixed;boundary=[any boundary name]

--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

IAM
Redirecting number
Address presentation restricted indicator
presentation allowed
Address signal (Diverting user)
Original called number
Address presentation restricted indicator
presentation allowed
Address signal
Redirection information
Original Redirection Reason
unknown
Redirecting indicator
Redirection counter
Redirecting reason
Mobile subscriber not reachable

--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A)   Interconnection Interface   SIP (Network B)
INVITE(Call-ID A-B) →
CFNL is performed
← INVITE(Call-ID B-C, IAM)
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A
Check: Is a INVITE request received at the interconnection interface?
Check: Is an IAM encapsulated in the INVITE?
Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'?
Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'?
Check: Is the Redirection number present?
Check: Is Redirection information present and the Redirecting reason is set to 'Mobile subscriber not reachable'?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_cfnl_017
Test case group	SIP-SIP/Service/CFNL
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CFNL performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Do not release diverting numberinformation
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Redirecting number Address presentation restricted indicator presentation restricted Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason Mobile subscriber not reachable --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CFNL is performed** ← INVITE(Call-ID B-C, IAM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check: Is a INVITE request received at the interconnection interface**? **Check:** Is an IAM encapsulated in the INVITE? **Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation restricted'? **Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation restricted'? **Check:** Is the Redirection number present? **Check:** Is Redirection information present and the Redirecting reason is set to 'Mobile subscriber not reachable'? Repeat this test in reverse direction.

7.1.5.6.5 Communication Deflection

Test case number	SS_cd_001
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 29
Test purpose	**Communication deflection during alerting, basic rules.** The user A and user C are in Network A. The user B is in network B and is provided with CDa. Ensure that when user A calls user B, the call is deflected during alerting to user C. In the active call state, ensure the property of speech.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CDa is performed		
	←	180 Ringing(Call-ID B-A)		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

Comments	**Check:** CDa is successful. **Check:** In the active call state, ensure the property of speech. **Check:** Is the P-Asserted-Identity present set to the identity of the originating user? Repeat this test in reverse direction.

Test case number	SS_cd_002
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 29
Test purpose	**Communication deflection immediate, basic rules.** Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C. In the active call state, ensure the property of speech.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CDi is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		

Comments	**Check:** CDi is successful. **Check:** In the active call state, ensure the property of speech. **Check:** Is the P-Asserted-Identity present set to the identity of the originating user? Repeat this test in reverse direction.

Test case number	SS_cd_003
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 29 AND SE 30
Test purpose	**Communication Deflection immediate response, no notification.** The user A and user C are in Network A. The user B is in network B and is provided with CFU, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded message.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = No
SIP Parameter	

Message flow

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CDi is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		Apply post test routine		

Comments	**Check:** No notification regarding call forwarding in network B is received at the interconnection interface. **Check:** Is the cause parameter in the last entry is set to '480'. Repeat this test in reverse direction.

Test case number	SS_cd_004
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 29 AND SE 30
Test purpose	**Communication Deflection immediate response, originating user is notified. URI of the diverted-to user not received.** The user A and user C are in network A. The user B is in network B and is provided with CFU Originating user receives notification that his communication has been diverted = Yes and ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message, user A is notified of call diversion and not informed of the diverted-to number and served user number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Originating user receives notification that his communication has been diverted = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No
SIP Parameter	181 Being Forwarded History-Info: <sip:userB@networkB?Privacy=history&Reason=SIP;cause=302>;index=1, <sip: userC@networkA;cause=480?Privacy=history>;index=1.1
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CDi is performed** ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) **Apply post test routine**
Comments	**Check:** A 181 Being Forwarded and a History-Info header is received at the interconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. **Check:** Is the cause parameter in the last entry is set to '480'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cd_005
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 29 AND SE 30
Test purpose	**Communication Deflection immediate response, originating user is notified. URI from the diverted-to user received.** The user A and user C are in network 1. The user B is in network N2 and is provided with CFU Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" =Yes. Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message, user A is notified of call diversion and informed of the diverted-to number.
Configuration	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of diverted to URI to originating user in diversion notification = Yes
SIP Parameter	181 Being Forwarded History-Info: <sip:userB@networkB?Reason=SIP;cause=302>;index=1, <sip: userC@networkA;cause=480>;index=1.1
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) ➔ **CDi is performed** ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) **Apply post test routine**
Comments	**Check:** A 181 Being Forwarded is received at the interconnection interface. **Check:** A History-Info header is contained in the 181 with the URI of the diverted-to user. **Check:** Is the cause parameter in the last entry is set to '480'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cd_006
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 29 AND SE 30
Test purpose	**Communication Deflection immediate response, diverted-to user does not receive the URI of the served user.** The user A and user C are in network A. The user B is in network B and is provided with CFU "Served user allows the presentation of his/her URI to the diverted-to user" = No. Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C, user C is not informed of the forwarding number.
Configuration	**Subscription options:** • Served user allows the presentation of his/her URI to diverted-to user = No
SIP Parameter	INVITE Request line contains ';cause=480' History-Info: <sip:userB@networkB?Privacy=history&Reason=SIP;cause=302>;index=1, <sip: userC@networkA;cause=480>;index=1.1
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CDi is performed** ← INVITE(Call-ID B-C) **Apply post test routine**
Comments	**Check:** A History-Info header is received in the INVITE contains the URI of user B (served user) at the interconnection interface and a Privacy header is escaped set to 'history'. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '480'. **Check:** Is the cause parameter in the last entry is set to '480'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cd_007
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 29 AND SE 30
Test purpose	**Communication Deflection immediate response, diverted-to user receives the URI of the served user.** The user A and user C are in network A. The user B is in network B and is provided with CFU "Served user allows the presentation of his/her URI to diverted-to user" = Yes. Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C, user C is informed of the forwarding number.
Configuration	**Subscription options:** • Served user allows the presentation of his/her URI to diverted-to user = Yes
SIP Parameter	INVITE Request line contains ';cause=480' History-Info: <sip:userB@networkB?Reason=SIP;cause=302>;index=1, <sip: userC@networkA;cause=480>;index=1.1
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CDi is performed** ← INVITE(Call-ID B-C) **Apply post test routine**
Comments	**Check:** A History-Info header is received in the INVITE contains the URI of user B (served user) at the interconnection interface. **Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '480'? **Check:** Is the cause parameter in the last entry is set to '480'? **NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS_cd_008
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 29
Test purpose	**Communication Deflection immediate response, unsuccessful UDUB.** The user A and user C are in network A. The user B is in network B and is provided with CDi. Ensure that when user A calls user B, the call is deflected immediate to user C user C is user determined user busy.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → **CDi is performed** ← INVITE(Call-ID B-C) 486 Busy Here(Call-ID C-B) → ← ACK(Call-ID B-C) ← 486 Busy Here(Call-ID B-A) ACK(Call-ID A-B) → **Apply post test routine**
Comments	**Check:** The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction.

Test case number	SS_cd_009
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 29
Test purpose	**Communication Deflection immediate response, unsuccessful NDUB.** The user A and user C are in network A. The user B is in network B. Ensure that when user A calls user B, the call is deflected immediate to user C and user C is network determined user busy.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CDi is performed		
	←	INVITE(Call-ID B-C)		
		486 Busy Here(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	486 Busy Here(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Apply post test routine		

Comments	**Check:** The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction

Test case number	SS_cd_010
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 29 AND SE 30 AND [Network A] SE 9
Test purpose	**Communication Deflection immediate response, interaction with a not trusted network.** The user A and user C are in network A. The user B is in network B and is provided with CD Originating user receives notification that his communication has been diverted = Yes ("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. Ensure that when user A calls user B, the call is deflected immediate response to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.
Configuration	
SIP Parameter	**Subscription options:** • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CDi is performed		
	←	INVITE(Call-ID B-C)		
	←	181 Being Forwarded(Call-ID B-A)		
		Apply post test routine		

Comments	**Check:** No History-Info header is received in the INVITE at the interconnection interface. **Check:** No History-Info header is received in the 181 Being Forwarded at the interconnection interface. Repeat this test in reverse direction.

Test case number	SS_cd_011
Test case group	SIP-SIP/Service/CD
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CD performed in Network B, Notification subscription options is set to presentation not allowed.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is deflected to user C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = no
SIP Parameter	183 /180 Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM/CPG** Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation not allowed Redirecting reason Deflection immediate or Deflection during alerting Generic notification call is diverting --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A, ACM) in case CDa **CD is performed** ← INVITE(Call-ID B-C, IAM) ← 183 / 180 (Call-ID B-A, ACM/CPG) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 183 Session Progress received at the interconnection interface? **Check:** Is an ACM encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator set to 'presentation not allowed'? **Check:** Is the Redirecting reason set to 'Deflection immediate' or 'Deflection during alerting'? Repeat this test in reverse direction.

Test case number	SS_cd_012
Test case group	SIP-SIP/Service/CD
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CD performed in Network B, Notification subscription options is set to presentation allowed without redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is deflected to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number
SIP Parameter	183 /180 Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM/CPG** Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed without redirection number Redirecting reason Deflection immediate or Deflection during alerting Generic notification call is diverting --[any boundary name]--
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
	←	180 Ringing (Call-ID B-A) in case CDa		
		CD is performed		
	←	INVITE(Call-ID B-C, IAM)		
	←	183 / 180 (Call-ID B-A, ACM/CPG)		
		Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** 183 Session Progress is received at the interconnection interface. **Check:** Is an ACM encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator is set to 'presentation allowed without redirection number'? **Check:** Is the Redirecting reason set to 'Deflection immediate' or 'Deflection during alerting'? Repeat this test in reverse direction.

Test case number	SS_cd_013
Test case group	SIP-SIP/Service/CD
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CD performed in Network B, Notification subscription options is set to presentation allowed with redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number. Ensure that when user A calls user B, the call is deflected to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	**Subscription options:** • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number
SIP Parameter	183 /180 Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM/CPG** Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed with redirection number Redirecting reason Deflection immediate or Deflection during alerting Generic notification call is diverting --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A) in case CDa **CD is performed** ← INVITE(Call-ID B-C, IAM) ← 183 / 180 (Call-ID B-A, ACM/CPG) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** 183 Session Progress is received at the interconnection interface. **Check:** Is an ACM encapsulated in the 183? **Check:** Is the Called party's status indicator set to 'no indication'? **Check:** Is the Redirection number present? **Check:** Is Notification subscription options indicator is set to 'presentation allowed with redirection number'? **Check:** Is the Redirecting reason set to 'Deflection immediate' or 'Deflection during alerting'? Repeat this test in reverse direction.

Test case number	SS_cd_014
Test case group	SIP-SIP/Service/CD
Reference	6.7/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. CD performed in Network B, Restriction of the Redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Diverted-to user is subscribed to the COLR service in Permanent mode. Ensure that when user A calls user B, the call is deflected to user C, a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	**Subscription options:** • Connected user subscribed to COLR, Permanent = yes
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** Redirection number restriction Presentation restricted --[any boundary name]--
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B), IAM	→	
	←	180 Ringing (Call-ID B-A) in case CDa		
		CD is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing (Call-ID C-B, ACM)	→	
	←	180 Ringing (Call-ID B-A)		
		200 OK INVITE (Call-ID C-B, ANM)	→	
	←	ACK (Call-ID B-C)		
	←	200 OK INVITE (Call-ID B-A)		
		ACK (Call-ID A-B)	→	
		Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 200 OK INVITE received at the interconnection interface? **Check:** Is an ANM encapsulated in the 200 OK? **Check:** Is the ISUP/BICC Redirection number restriction set to 'Presentation restricted'? Repeat this test in reverse direction.

Test case number	SS_cd_015
Test case group	SIP-SIP/Service/CD
Reference	6.7/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	**SIP-I support. CD performed in Network B, No restriction of the Redirection number.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Diverted-to user is not subscribed to the COLR service. Ensure that when user A calls user B, the call is deflected to user C, if a Redirection number restriction parameter is present it is set to 'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	**Subscription options:** • Connected user subscribed to COLR = no
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** Redirection number restriction Presentation allowed or Redirection number restriction not present --[any boundary name]--
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE(Call-ID A-B), IAM	→	
	←	180 Ringing (Call-ID B-A) in case CDa		
		CD is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing (Call-ID C-B, ACM)	→	
	←	180 Ringing (Call-ID B-A)		
		200 OK INVITE (Call-ID C-B, ANM)	→	
	←	ACK (Call-ID B-C)		
	←	200 OK INVITE (Call-ID B-A)		
		ACK (Call-ID A-B)	→	
		Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** Is a 200 OK INVITE received at the interconnection interface? **Check:** Is an ANM encapsulated in the 200 OK? **Check:** Is the ISUP/BICC Redirection number restriction present set to 'Presentation allowed' or is the parameter absent? Repeat this test in reverse direction.

Test case number	SS_cd_016
Test case group	SIP-SIP/Service/CD
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	**SIP-I support. CD performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.** The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is deflected to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.
Configuration	**Subscription options:** • Served user releases his/her number to diverted-to user = Release diverting number information
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Redirecting number Address presentation restricted indicator presentation allowed Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason Deflection immediate or Deflection during alerting --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A) in case CDa **CD is performed** ← INVITE(Call-ID B-C, IAM) **Apply post test routine**
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A **Check:** **Is a INVITE request received at the interconnection interface**? **Check:** Is an IAM encapsulated in the INVITE? **Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? **Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? **Check:** Is the Redirection number present? **Check:** Is Redirection information present and the Redirecting reason is set to 'Deflection immediate' or 'Deflection during alerting'? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_cd_017</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CD</td></tr>
<tr><td>Reference</td><td>7.1/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network B] SE 17 AND SE 47 AND SE 55</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. CD performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.

The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Served user releases his/her number to diverted-to user = Release diverting number information.
Ensure that when user A calls user B, the call is deflected to user C, user C is notified of call diversion and informed of the diverting number.
The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.</td></tr>
<tr><td>Configuration</td><td>Subscription options:
• Served user releases his/her number to diverted-to user = Do not release diverting numberinformation</td></tr>
<tr><td>SIP Parameter</td><td>INVITE
Content-Type: multipart/mixed;boundary=[any boundary name]

--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

IAM
Redirecting number
Address presentation restricted indicator
presentation restricted
Address signal (Diverting user)
Original called number
Address presentation restricted indicator
presentation restricted
Address signal
Redirection information
Original Redirection Reason
unknown
Redirecting indicator
Redirection counter
Redirecting reason
Deflection immediate or Deflection during alerting

--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →
← 180 Ringing (Call-ID B-A) in case CDa
CD is performed
← INVITE(Call-ID B-C, IAM)
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A
Check: Is a INVITE request received at the interconnection interface?
Check: Is an IAM encapsulated in the INVITE?
Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation restricted'?
Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation restricted'?
Check: Is the Redirection number present?
Check: Is Redirection information present and the Redirecting reason is set to 'Deflection immediate' or 'Deflection during alerting'?
Repeat this test in reverse direction.</td></tr>
</table>

7.1.5.7 Conference (CONF)

Test case number	SS_conf_001
Test case group	SIP-SIP/Service/CONF
Reference	4.5.2/[10]
SELECTION EXPRESSION	([Network A] SE 11 AND [Network B] SE 11) AND SE 31
Test purpose	**3 Party establishment using the REFER method.** User B1 and user B2 are located in network B, user A is located in network A. A confirmed session from user A to user B1 is set on hold; a confirmed session from user A to user B2 is set on hold. • Ensure that when user A refers to user B1 to invite to the conference, the user B1 sends a NOTIFY to user A indicating 'Tying'. The user B1 sends an INVITE request to the conference focus in network A. Is the request is confirmed, user B1 sends a NOTIFY indicating '200 OK'. User A terminates the original dialogue. • Ensure that when user A refers to user B2 to invite to the conference, the user B2 sends a NOTIFY to user A indicating 'Tying'. The user B2 sends an INVITE request to the conference focus in network A. Is the request is confirmed, user B2 sends a NOTIFY indicating '200 OK'. User A terminates the original dialogue.
Configuration	
SIP Parameter	REFER(user B1) Refer-To: <uri of conference focus;method=INVITE > NOTIFY(B1, 100) Content-Type: message/sipfrag SIP/2.0 100 INVITE: Request URI: uri of conference focus From: user B1 NOTIFY(B1, 200) Content-Type: message/sipfrag SIP/2.0 200 OK REFER(user B2) Refer-To: <uri of conference focus;method=INVITE > NOTIFY(B2, 100) Content-Type: message/sipfrag SIP/2.0 100 INVITE: Request URI: uri of conference focus From: user B2 NOTIFY(B2, 200) Content-Type: message/sipfrag SIP/2.0 200 OK

Message flow

SIP (Network A)	**Interconnection Interface**	**SIP (Network B)**
	Establish a confirmed session to user B1 from Network A to Network B and put it on hold	
	Establish a confirmed session to user B2 from Network A to Network B and put it on hold	
	User A establishes a 3PTY conversation	
	REFER(user B1)	→
←	202 Accepted	
←	NOTIFY(B1, 100)	
	200 OK NOTIFY	→
←	INVITE(focus, user B1)	
	200 INVITE	→
←	ACK	
←	NOTIFY(B1, 200)	
	200 OK NOTIFY	→
	BYE(user B1)	→
←	200 OK BYE	
	REFER(user B2)	→
←	202 Accepted	
←	NOTIFY(100)	
	200 OK NOTIFY	→
←	INVITE(focus, user B2)	
	200 INVITE	→
←	ACK	
←	NOTIFY(B2, 200)	
	200 OK NOTIFY	→
	BYE(user B2)	→
←	200 OK BYE	
	Apply post test routine	

Comments	User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold **Check:** The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter set to 'INVITE'. **Check:** The NOTIFY after the REFER request contains the 'SIP/2.0 100' message body. **Check:** The INVITE request is sent by user B1 and user B2 to the conference focus the Request URI is used from the Refer-To header of the received REFER request. **Check:** The NOTIFY after the REFER request contains the 'SIP/2.0 200 OK' message body. **Check:** The original session is terminated by user A. Repeat this test in reverse direction.

Test case number	SS_conf_002
Test case group	SIP-SIP/Service/CONF
Reference	4.5.2/[10], 4.7.2.9.7/[20]
SELECTION EXPRESSION	[Network A] SE 12 AND SE 31
Test purpose	**3 Party establishment using reINVITE performed by the AS in network A.** User B1 and user B2 are located in network B, user A is located in network A. A confirmed session from user A to user B1 is set on hold; a confirmed session from user A to user B2 is set on hold. • Ensure that user A can invite user B1 to the conference by sending a reINVITE request. • Ensure that user A can invite user B2 to the conference by sending a reINVITE request.
Configuration	
SIP Parameter	INVITE <B1> From: <userA> To: <userB1> Call-ID: A-B1 P-Asserted-Identity: <userA> SDP: a=sendrecv INVITE <B2> From: <userA> Call-ID: A-B2 To: <userB2> P-Asserted-Identity: <userA> SDP: a=sendrecv
Message flow	**SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **Establish a confirmed session to user B1 from Network A to Network B and put it on hold** **Establish a confirmed session to user B2 from Network A to Network B and put it on hold** **User A establishes a 3PTY conversation** INVITE(Call-ID A-B1) → ← 200 INVITE ACK → INVITE(Call-ID A-B2) → ← 200 INVITE ACK → **Apply post test routine**
Comments	**User** A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold **Check:** An INVITE is sent to user B1 and user B2 indicating a new IP address in the 'c' line of the SDP. **Check:** The 'a' line indicates 'sendrecv'. Repeat this test in reverse direction.

Test case number	SS_conf_003
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	**SIP-I/ISUP interworking. Served user establishes a 3 Party communication.** Served User A is located in Network A and ISUP/BICC - SIP-I interworking applies in Network A. User A establishes a confirmed communication with a User B1 in Network B and sets it on HOLD. User A establishes a confirmed communication with a User B2 in Network B. Ensure that when User A establishes a 3 PTY communication: • an INFO request is sent to User B1 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established'; • an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established'.
Configuration	ISUP/BICC interworking applies in Network A User in Network A is subscribed to the 3PTY supplementary service
SIP Parameter	INFO <B1> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic Notification Conference established INFO <B2> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic Notification Conference established
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **Establish a confirmed session from User A in Network A to user B1 in Network B and put it on hold** **Establish a confirmed session from User A in Network A to user B2 in Network B** INFO(Call-ID A-B1, CPG) → ← 200 INFO INFO(Call-ID A-B2, CPG) → ← 200 INFO **Apply post test routine**	
Comments	**User A** establishes confirmed communication to user B1 in Network B and sets it on hold **User A** establishes a confirmed communication to user B2 in Network B **User A** invokes the 3PTY communication **Check:** Is an INFO request sent to user B1 and user B2 in Network B? **Check:** Is a ISUP/BICC CPG message encapsulated in the INFO request to both remote users in Network B? **Check:** Is the Generic Notification parameter in the encapsulated CPG in both INFO set to 'Conference established'? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_conf_004</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CONF</td></tr>
<tr><td>Reference</td><td>5.4/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 17 AND SE 47 AND SE 56</td></tr>
<tr><td>Test purpose</td><td>SIP-I/ISUP interworking. Served user disconnects one of the remote users.

Served User A is located in Network A and ISUP/BICC - SIP-I interworking applies in Network A. User A establishes a confirmed communication with a User B1 in Network B and sets it on HOLD. User A establishes a confirmed communication with a User B2 in Network B. User A invokes 3PTY conversation. Ensure that when User A disconnects the previous active user:
• a BYE request is sent to User B1 in Network B;
• an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Conference disconnected'.</td></tr>
<tr><td>Configuration</td><td>ISUP/BICC interworking applies in Network A
User in Network A is subscribed to the 3PTY supplementary service</td></tr>
<tr><td>SIP Parameter</td><td>INFO <B2>
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

CPG
Generic Notification
Conference disconnected</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
Establish a confirmed session from User A in Network A to user B1 in Network B and put it on hold
Establish a confirmed session from User A in Network A to user B2 in Network B
User A establishes a 3PTY conversation
BYE(Call-ID A-B1, REL) →
← 200 INFO

INFO(Call-ID A-B2, CPG) →
← 200 INFO
Apply post test routine</td></tr>
<tr><td>Comments</td><td>User A establishes a 3PTY conversation with user B1 and user B2 located in Network B
User A disconnects the communication with user B1 in Network B (previous on hold)
Check: Is a BYE request is sent to user B1 in Network B?
Check: Is a ISUP/BICC CPG message encapsulated in the INFO request to user B2 in Network B?
Check: Is the Generic Notification parameter in the encapsulated CPG in the INFO sent to user B2 set to 'Conference disconnected'?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_conf_005
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	**SIP-I/ISUP interworking. Served user splits the 3 Party communication.** Served User A is located in Network A and ISUP/BICC - SIP-I interworking applies in Network A. User A establishes a confirmed communication with a User B1 in Network B and sets it on HOLD. User A establishes a confirmed communication with a User B2 in Network B. User A invokes 3PTY conversation Ensure that when User A splits the 3 PTY communication: • an INFO request is sent to User B1 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Conference disconnected'; • an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Conference disconnected'.
Configuration	ISUP/BICC interworking applies in Network A User in Network A is subscribed to the 3PTY supplementary service
SIP Parameter	INFO <B1> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic Notification Conference disconnected INFO <B2> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic Notification Conference disconnected
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **Establish a confirmed session from User A in Network A to user B1 in Network B and put it on hold** **Establish a confirmed session from User A in Network A to user B2 in Network B** **User A establishes a 3PTY conversation** INFO(Call-ID A-B1, CPG) → ← 200 INFO INFO(Call-ID A-B2, CPG) → ← 200 INFO **Apply post test routine**	
Comments	**User A** establishes confirmed communication to user B1 in Network B and sets it on hold **User A** establishes a confirmed communication to user B2 in Network B **Check:** Is an INFO request sent to user B1 and user B2 in Network B? **Check:** Is a ISUP/BICC CPG message encapsulated in the INFO request to both remote users in Network B? **Check:** Is the Generic Notification parameter in the encapsulated CPG in both INFO set to 'Conference established'? Repeat this test in reverse direction.

Test case number	SS_conf_006
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	**SIP-I/ISUP interworking. Establishment of aCONF conversation.** Served User A is located in Network A and ISUP/BICC - SIP-I interworking applies in Network A. User A establishes a confirmed communication with a User B1 in Network B and invokes the CONF communication. Ensure that when User A invokes the CONF communication: • an INFO request is sent to User B1 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established' when the conference is invoked. User A establishes a confirmed communication with a User B2 in Network B. Ensure when User A adds the user B2 to the established conference: • an INFO request is sent to User B1 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Other party; • an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established' when the user is added to the conference.
Configuration	ISUP/BICC interworking applies in Network A User in Network A is subscribed to the 3PTY supplementary service
SIP Parameter	INFO1 <B1> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic Notification conference established INFO2 <B1> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic Notification Other party added INFO <B2> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic Notification conference established
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **Establish a confirmed session from User A in Network A to user B1 in Network B** **User A establishes a CONF conversation** INFO1(Call-ID A-B1, CPG) → ← 200 INFO **Establish a confirmed session from User A in Network A to user B2 in Network B and add to the conference** INFO2(Call-ID A-B2, CPG) → ← 200 INFO INFO(Call-ID A-B2, CPG) → ← 200 INFO **Apply post test routine**	

Comments	**User A** establishes confirmed communication to user B1 in Network B and invoke the CONF communication **Check:** Is an INFO request sent to user B1 and in Network B and Is a ISUP/BICC CPG message encapsulated in the INFO request and the Generic Notification is set to '**conference established**'? **User A** establishes a confirmed communication to user B2 in Network B and add it to the conference. **Check:** Is an INFO request sent to user B2 Network B and a ISUP/BICC CPG message encapsulated the Generic Notification is set to '**conference established**'? **Check:** Is an INFO request sent to user B1 Network B and a ISUP/BICC CPG message encapsulated the Generic Notification is set to '**Other party added**'? Repeat this test in reverse direction.

Test case number	SS_conf_007
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	**SIP-I/ISUP interworking. Isolation and Reattachment of one party of the conference.** Served User A is located in Network A and ISUP/BICC - SIP-I interworking applies in Network A. User A invokes a CONF communication with user B1 and user B2 in Network B. Ensure that when User A isolates one remote party (B1) from the CONF communication: • an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'isolated' in the encapsulated ISUP/BICCCPG; • an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party isolated' in the encapsulated ISUP/BICCCPG. Ensure that when User A reattaches one remote party (B1) to the CONF communication: • an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'reattached' in the encapsulated ISUP/BICCCPG; • an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party reattached' in the encapsulated ISUP/BICCCPG.
Configuration	ISUP/BICC interworking applies in Network A User in Network A is subscribed to the 3PTY supplementary service
SIP Parameter	INFO1 <B1> **CPG** Generic Notification= isolated INFO2 <B1> **CPG** Generic Notification= Other party isolated INFO1 <B2> **CPG** Generic Notification= reattached INFO2 <B2> **CPG** Generic Notification= Other party reattached

Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **Establish a CONF communication with User B1 and User B2 in Network B** **User A isolates User B1 from the CONF conversation** INFO1(Call-ID A-B1, CPG) → ← 200 INFO INFO1(Call-ID A-B2, CPG) → ← 200 INFO **User A reattaches User B1 to the CONF conversation** INFO2(Call-ID A-B2, CPG) → ← 200 INFO INFO2(Call-ID A-B2, CPG) → ← 200 INFO **Apply post test routine**	
Comments	**User A** Invokes a CONF conversation with User B1 and User b2 in Network B **User A** splits user B1 in Network B from the CONF conversation **Check:** Is an INFO request sent to user B1 and the Generic notification is set to 'isolated' in the encapsulated CPG? **Check:** Is an INFO request sent to user B2 and the Generic notification is set to 'Other party isolated' in the encapsulated CPG? **User A** reattaches user B1 in Network B to the CONF conversation **Check:** Is an INFO request sent to user B1 and the Generic notification is set to 'reattached' in the encapsulated CPG? **Check:** Is an INFO request sent to user B2 and the Generic notification is set to 'Other party reattached' in the encapsulated CPG? Repeat this test in reverse direction

Test case number	SS_conf_008
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	**SIP-I/ISUP interworking. Splitting and Adding of a party.** Served User A is located in Network A and ISUP/BICC - SIP-I interworking applies in Network A. User A invokes a CONF communication with user B1 and user B2 in Network B. Ensure that when User A split one remote party (B1) from the CONF communication: • an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'conference disconnected' in the encapsulated ISUP/BICCCPG; • an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party split' in the encapsulated ISUP/BICCCPG. Ensure that when User A adds one remote party (B1) to the CONF communication: • an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'Conference established' in the encapsulated ISUP/BICCCPG; • an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party added' in the encapsulated ISUP/BICCCPG.
Configuration	ISUP/BICC interworking applies in Network A User in Network A is subscribed to the 3PTY supplementary service
SIP Parameter	INFO1 <B1> **CPG** Generic Notification= conference disconnected INFO2 <B1> **CPG** Generic Notification=Other party split INFO1 <B2> **CPG** Generic Notification=Conference established INFO2 <B2> **CPG** Generic Notification= Other party added
Message flow	
SIP (Network A) **Interconnection Interface** **SIP (Network B)** **Establish a CONF communication with User B1 and User B2 in Network B** **User A isolates User B1 from the CONF conversation** INFO1(Call-ID A-B1, CPG) → ← 200 INFO INFO1(Call-ID A-B2, CPG) → ← 200 INFO **User A reattaches User B1 to the CONF conversation** INFO2(Call-ID A-B2, CPG) → ← 200 INFO INFO2(Call-ID A-B2, CPG) → ← 200 INFO **Apply post test routine**	
Comments	**User A** Invokes a CONF conversation with User B1 and User b2 in Network B **User A** splits user B1 in Network B from the CONF conversation. **Check:** Is an INFO request sent to user B1 and the Generic notification is set to 'conference disconnected' in the encapsulated CPG? **Check:** Is an INFO request sent to user B2 and the Generic notification is set to 'Other party split' in the encapsulated CPG? **User A** adds user B1 in Network B to the CONF conversation. **Check:** Is an INFO request sent to user B1 and the Generic notification is set to 'Conference established' in the encapsulated CPG? **Check:** Is an INFO request sent to user B2 and the Generic notification is set to 'Other party added' in the encapsulated CPG? Repeat this test in reverse direction

7.1.5.8 Anonymous Communication Rejection (ACR) and Communication Barring (CB)

Test case number	SS_acr-cb_001
Test case group	SIP-SIP/Service/ACR-CB
Reference	4.5.2.6/[12]
SELECTION EXPRESSION	SE 32
Test purpose	**Call Barring performed in network B for user B.** User A is located in network A and user B is located in network B and is subscribed to the Incoming Call Barring service. Ensure that a communication from user A is rejected in network B by sending a 603 Decline due to the Call Barring service of user B.
Configuration	User B is subscribed to the incoming Call Barring service (e.g. user A in a black list)
SIP Parameter	INVITE P-Asserted-Identity: <URI of user A>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 603 (Decline) ACK →
Comments	**Check:** Is the P-Asserted-Identity present? **Check:** Is the communication rejected by sending a 603 (Decline) final response sent to user A? Repeat this test in reverse direction.

Test case number	SS_acr-cb_002
Test case group	SIP-SIP/Service/ACR-CB
Reference	4.5.2.6/[12]
SELECTION EXPRESSION	SE 33
Test purpose	**ACR performed in network B for user B.** User A is located in network A and user B is located in network B and is subscribed to the Anonymous Communication rejection service. Ensure that an anonymous communication from user A is rejected in network B by sending a 403 Anonymity Disallowed final response due to the Anonymous Communication Rejection service of user B.
Configuration	User B is subscribed to the Anonymous Communication Rejection service
SIP Parameter	INVITE P-Asserted-Identity: <URI of user A> Privacy: id
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 433 (Anonymity Disallowed) ACK →
Comments	**Check:** Is the P-Asserted-Identity present? **Check:** Is the Privacy header set to 'id'? **Check:** Is the communication rejected by sending a 433 (Anonymity Disallowed) final response sent to user A? Repeat this test in reverse direction.

Test case number	SS_acr-cb_003
Test case group	SIP-SIP/Service/ACR-CB
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 57
Test purpose	**SIP-I interworking. ACR performed in network B for user B.** User A is located in network A and user B is in the PSTN/PLMN part of Network B and is subscribed to the Anonymous Communication rejection service. Ensure that an anonymous communication from user A is rejected in network B by sending a 603 Decline final response due to the Anonymous Communication Rejection service of user B. A ISUP/BICC REL is present in the 603 the Cause indicator value is set to '21' if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	User B is subscribed to the Anonymous Call Rejection service
SIP Parameter	INVITE P-Asserted-Identity: <URI of user A> Privacy: id 433 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause = 21
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE → ← 603 Decline (REL) ACK →	
Comments	**Check:** Is the P-Asserted-Identity present? **Check:** Is the Privacy header set to 'id'? **Check:** Is the communication rejected by sending a 603 Decline final response sent to user A? **Check:** Is an ISUP/BICC REL is present in the 603 and the cause value is set to '21'? Repeat this test in reverse direction.

7.1.5.9 Closed User Group (CUG)

Test case number	SS_cug_001
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.4/[13]
SELECTION EXPRESSION	SE 34
Test purpose	**Originating user +OA to terminating user no CUG.** An originating user in a CUG Outgoing Access allowed calls to a user not in a CUG. The session establishment is successful.
Configuration	**Originating user**: CUG, outgoing access allowed
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml ……. <…cug> <…networkIndicator>01</… networkIndicator <…networkIndicator>23</… networkIndicator <….cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode> <…CugCommunicationIndicator>10</…cugCommunicationIndicator> <…:cug>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing **Apply post test routine**
Comments	**Check:** Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml? **Check:** Contains the XML body in the INVITE a 'cug' element? **Check:** Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element? **Check:** Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element? **Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '10' as a 'cug' child element? **Check:** Is the session setup not rejected? Repeat this test in reverse direction. **NOTE:** The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug_002
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.4, 4.5.2.10/[13]
SELECTION EXPRESSION	SE 34
Test purpose	**Originating user -OA to terminating user no CUG.** An originating user in a CUG Outgoing Access not allowed calls to a user not in a CUG. The session establishment is not successful, a 403 (Forbidden) response is sent.
Configuration	**Originating user**: CUG, outgoing access not allowed
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml Content-Disposition: ….;handling= required ……. <…:cug> <…networkIndicator>01</…networkIndicator <…networkIndicator>23</…networkIndicator <….ugInterlockBinaryCode>0F03</…cugInterlockBinaryCode> <…cugCommunicationIndicator>11</…cugCommunicationIndicator> <…cug>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 403 (Forbidden) ACK →
Comments	**Check:** Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml? **Check:** Is the handling parameter in the Content-Disposition header set to required? **Check:** Contains the XML body in the INVITE a 'cug' element? **Check:** Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element? **Check:** Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element? **Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? **Check:** Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network? Repeat this test in reverse direction. **NOTE:** The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug_003
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.4, 4.5.2.10/[13]
SELECTION EXPRESSION	SE 34
Test purpose	**Originating user -OA to terminating user -IA.** An originating user in a CUG Outgoing Access not allowed calls to a user in the same CUG Incoming Access not allowed. The session establishment is successful.
Configuration	**Originating user**: CUG, outgoing access not allowed **Terminating user**: CUG incoming access not allowed User in network A and user in network B are in the same CUG
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml Content-Disposition: ….;handling= required ……. <…cug> <…networkIndicator>01</…networkIndicator <…networkIndicator>23</…networkIndicator <…cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode> <…cugCommunicationIndicator>11</…cugCommunicationIndicator> <…cug>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE ➔ 🡐 180 Ringing **Apply post test routine**
Comments	**Check:** Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml? **Check:** Is the handling parameter in the Content-Disposition header set to required? **Check:** Contains the XML body in the INVITE a 'cug' element? **Check:** Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element? **Check:** Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element? **Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? **Check:** Is the session setup not rejected? Repeat this test in reverse direction. **NOTE:** The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug_004
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.4, 4.5.2.10/[13]
SELECTION EXPRESSION	SE 34
Test purpose	**Originating user in a CUG to terminating user -IA.** An originating user in a CUG calls to a user in a different CUG Incoming Access not allowed. The session establishment is not successful, a 403 (Forbidden) response is sent.
Configuration	User in network A and user in network B are not in the same CUG **Terminating user:** CUG incoming access not allowed
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml Content-Disposition: ….;handling= requiredv ……. <…cug> <…networkIndicator>01</…networkIndicator <…networkIndicator>23</…networkIndicator <…cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode> <…cugCommunicationIndicator>..</…cugCommunicationIndicator> <…cug>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 403 (Forbidden) ACK →
Comments	**Check:** Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml? **Check:** Contains the XML body in the INVITE a 'cug' element? **Check:** Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element? **Check:** Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element? **Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '10' or '11'as a 'cug' child element? **Check:** Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network? Repeat this test in reverse direction. **NOTE:** The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug_005
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.10/[13]
SELECTION EXPRESSION	SE 34
Test purpose	**Originating user no CUG to terminating user +IA.** An originating user not in a CUG calls to a user in a CUG Incoming Access allowed. The session establishment is successful.
Configuration	**Terminating user**: CUG incoming access allowed
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing **Apply post test routine**
Comments	**Check:** Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network. Repeat this test in reverse direction.

Test case number	SS_cug_006
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.10/[13]
SELECTION EXPRESSION	[Network A] SE 34 AND NOT [Network B] SE 34
Test purpose	**Originating user no CUG to terminating user -IA.** An originating user not in a CUG calls to a user in a CUG Incoming Access not allowed. The session establishment is not successful, a 403 (Forbidden) response is sent.
Configuration	User in Network B in a CUG incoming access not allowed
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 403 (Forbidden) ACK →
Comments	**Check:** Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network. Repeat this test in reverse direction.

Test case number	SS_cug_007
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.4/[13]
SELECTION EXPRESSION	SE 34
Test purpose	**Originating user -OA, network B does not support CUG.** An originating user in a CUG Outgoing Access not allowed calls to a user in network B. Network B does not support CUG. The session establishment is not successful, a 4xx unsuccessful final response is sent.
Configuration	
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml Content-Disposition: ….;handling= required ……. <…cug> <…networkIndicator>01</…networkIndicator <…networkIndicator>23</…networkIndicator <…cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode> <…cugCommunicationIndicator>10</…cugCommunicationIndicator> <…cug>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 4xx/501 Not Implemented ACK →
Comments	**Check:** Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml? **Check:** Is the handling parameter in the Content-Disposition header set to required? **Check:** Contains the XML body in the INVITE a 'cug' element? **Check:** Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element? **Check:** Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element? **Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? **Check:** Is the session setup rejected by sending an unsuccessful final response? Repeat this test in reverse direction. **NOTE:** The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug_008
Test case group	SIP-SIP/Service/CUG
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 58
Test purpose	**SIP-I/ISUP interworking. CUG call with outgoing access allowed.** User A is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. ensure that when user A is in a CUG 'outgoing access allowed' calls user B in Network B. The call is successful. There is a Optional forward call indicator the CUG Call Indicator **Outgoing access allowed** present in the encapsulated IAM sent to Network B.
Configuration	• User in PSTN/PLMN part of Network A in a CUG outgoing access allowed
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** **Optional Forward call indicator** CUG Call Indicator Outgoing access allowed **CUG interlock code** --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing
Comments	**User A** in the PSTN part of Network A calls user B in Network B **Check:** Is an IAM encapsulated in the INVITE request sent from Network A to Network B? **Check:** Is the Optional forward call indicator present, the CUG Call Indicator is set to 'Outgoing access allowed'? **Check:** Is the CUG interlock code parameter present in the encapsulated IAM? **NOTE:** CUG outgoing access allowed can appear like a basic call. Repeat this test in reverse direction.

Test case number	SS_cug_009
Test case group	SIP-SIP/Service/CUG
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 58
Test purpose	**SIP-I/ISUP interworking. CUG call with outgoing access not allowed.** User A is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. ensure that when user A is in a CUG 'outgoing access allowed' calls user B in Network B. The call is successful. There is a Optional forward call indicator the CUG Call Indicator **Outgoing access not allowed** present in the encapsulated IAM sent to Network B.
Configuration	• User in PSTN/PLMN part of Network A in a CUG outgoing access not allowed
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** **Optional Forward call indicator** CUG Call Indicator Outgoing access not allowed **CUG interlock code** --[any boundary name]--
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing	
Comments	**User A** in the PSTN part of Network A calls user B in Network B **Check:** Is an IAM encapsulated in the INVITE request sent from Network A to Network B? **Check:** Is the Optional forward call indicator present, the CUG Call Indicator is set to 'Outgoing access not allowed'? **Check:** Is the CUG interlock code parameter present in the encapsulated IAM? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_cug_010</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CUG</td></tr>
<tr><td>Reference</td><td>7.1/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>([Network A] SE 17 AND SE 47 AND SE 58) AND ([Network B] SE 17 AND SE 47 AND SE 58)</td></tr>
<tr><td>Test purpose</td><td>SIP-I/ISUP interworking. CUG call with outgoing access not allowed (both user in the same CUG).

User A in a CUG is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. User B is located in the PSTN/PLMN part and SIP-I - ISUP/BICC interworking applies in the same CUG. Ensure that when user A is in a CUG 'outgoing access not allowed' calls user B in Network B. The call is successful. There is a Optional forward call indicator the CUG Call Indicator Outgoing access not allowed present in the encapsulated IAM sent to Network B.</td></tr>
<tr><td>Configuration</td><td>• User in PSTN/PLMN part of Network A in a CUG outgoing access not allowed
• User in PSTN/PLMN part of Network B in a CUG
• User A and User B are in the same CUG</td></tr>
<tr><td>SIP Parameter</td><td>INVITE
Content-Type: multipart/mixed;boundary=[any boundary name]

--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

IAM
Optional Forward call indicator
CUG Call Indicator
Outgoing access not allowed
CUG interlock code
--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A)   Interconnection Interface   SIP (Network B)
INVITE →
← 180 Ringing</td></tr>
<tr><td>Comments</td><td>User A in the PSTN part of Network A calls user B in the PST/PLMN part of Network B
Check: Is an IAM encapsulated in the INVITE request sent from Network A to Network B?
Check: Is the Optional forward call indicator present, the CUG Call Indicator is set to 'Outgoing access not allowed'?
Check: Is the CUG interlock code parameter present in the encapsulated IAM?
Check: Is the call setup successful?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_cug_011
Test case group	SIP-SIP/Service/CUG
Reference	7.1/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 58) AND ([Network B] SE 17 AND SE 47 AND SE 58)
Test purpose	**SIP-I/ISUP interworking. CUG call to a CUG user incoming access not allowed (both user in the same CUG).** User A in a CUG is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. User B is located in the PSTN/PLMN part and SIP-I - ISUP/BICC interworking applies in the same CUG. Ensure that when user A is in a CUG 'outgoing access not allowed' calls CUG user B in Network B. The call is successful. There is a Optional forward call indicator the CUG Call Indicator **Outgoing access not allowed** present in the encapsulated IAM sent to Network B.
Configuration	• User in PSTN/PLMN part of Network A in a CUG outgoing access not allowed • User in PSTN/PLMN part of Network B in a CUG incoming access not allowed • User A and User B are in the same CUG
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** **Optional Forward call indicator** CUG Call Indicator Outgoing access not allowed **CUG interlock code** --[any boundary name]--
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing
Comments	**User A** in the PSTN/PLMN part of Network A calls user B in Network B **User B** in the PSTN/PLMN part of Network B. **Check:** Is an IAM encapsulated in the INVITE request sent from Network A to Network B? **Check:** Is the Optional forward call indicator present, the CUG Call Indicator is set to 'Outgoing access not allowed'? **Check:** Is the CUG interlock code parameter present in the encapsulated IAM? **Check:** Is the call setup successful? Repeat this test in reverse direction.

Test case number	SS_cug_012
Test case group	SIP-SIP/Service/CUG
Reference	7.1/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 58) AND ([Network B] SE 17 AND SE 47 AND SE 58)
Test purpose	**SIP-I/ISUP interworking. CUG call to a CUG user incoming access not allowed (both user in different CUG).** User A in a CUG is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. User B is located in the PSTN/PLMN part and SIP-I - ISUP/BICC interworking applies in different CUG. Ensure that when user A is in a CUG 'outgoing access not allowed' calls CUG user B in Network B. There is a Optional forward call indicator the CUG Call Indicator **Outgoing access not allowed** present in the encapsulated IAM sent to Network B. The call is rejected with a 500 (Server Internal error) final response. A ISUP/BICC REL is encapsulated and the Cause value is set to '87'.
Configuration	• User in PSTN/PLMN part of Network A in a CUG outgoing access not allowed • User in PSTN/PLMN part of Network B in a CUG incoming access not allowed • User A and User B are in different CUG
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** **Optional Forward call indicator** CUG Call Indicator Outgoing access not allowed **CUG interlock code** --[any boundary name]-- 500 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **REL** **Cause indicators** Cause value 87
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 500 Server Internal error(REL) ACK →
Comments	**User A** in the PSTN/PLMN part of Network A calls user B in Network B **User B** in the PSTN/PLMN part of Network B. **Check:** Is an IAM encapsulated in the INVITE request sent from Network A to Network B? **Check:** Is the Optional forward call indicator present, the CUG Call Indicator is set to 'Outgoing access not allowed'? **Check:** Is the CUG interlock code parameter present in the encapsulated IAM? **Check:** Is the call rejected with a 500 final response and a ISUP/BICC REL is encapsulated and the cause value is set to 87? Repeat this test in reverse direction.

Test case number	SS_cug_013
Test case group	SIP-SIP/Service/CUG
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 58
Test purpose	**SIP-I/ISUP interworking. Call to a CUG user incoming access not allowed.** User A is located in Network A. User B in a CUG Incoming access not allowed is located in the PSTN/PLMN part and SIP-I - ISUP/BICC interworking applies. Ensure that when user A calls user B in Network B. The call is rejected with a 500 (Server Internal error) final response. A ISUP/BICC REL is encapsulated and the Cause value is set to '87'.
Configuration	• User in PSTN/PLMN part of Network B in a CUG incoming access not allowed
SIP Parameter	500 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **REL** **Cause indicators** Cause value 87
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 500 Server Internal error(REL) ACK →
Comments	**User A** in Network A calls user B in Network B **User B** in the PSTN/PLMN part of Network B. **Check:** Is the call rejected with a 500 final response and a ISUP/BICC REL is encapsulated and the cause value is set to 87? Repeat this test in reverse direction.

Test case number	SS_cug_014
Test case group	SIP-SIP/Service/CUG
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 58
Test purpose	**SIP-I/ISUP interworking. Call to a CUG user incoming access allowed.** User A is located in Network A. User B is located in the PSTN/PLMN part and SIP-I - ISUP/BICC interworking applied. Ensure that when user A calls CUG user B Incoming access allowed in Network B. The call is successful.
Configuration	• User in PSTN/PLMN part of Network B in a CUG incoming access allowed
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing
Comments	**User A** in Network A calls user B in Network B **User B** in the PSTN/PLMN part of Network B. **Check:** Is the call setup successful? Repeat this test in reverse direction.

7.1.5.10 Communication Waiting (CW)

Test case number	SS_cw_001
Test case group	SIP-SIP/Service/CW
Reference	4.5.5.2/[15]
SELECTION EXPRESSION	SE 35
Test purpose	**Call Waiting indication in 180 response.** User A is located in network A, user B is located in network B and subscribed to the communication Waiting service. Ensure that when user A calls user B, user A receives the 'communication Waiting indication' in the 180 Ringing provisional response if the user B is NDUB or UDUB.
Configuration	User B subscribed to the CW service
SIP Parameter	180: Alert-Info: <urn:alert:service:call-waiting>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing **Apply post test routine**
Comments	**Check:** Is an Alert-Info header present in the 180 Ringing Response and is the value set to '<urn:alert:service:call-waiting>'? Repeat this test in reverse direction.

Test case number	SS_cw_002
Test case group	SIP-SIP/Service/CW
Reference	4.5.5.2/[15]
SELECTION EXPRESSION	SE 35 AND SE 36
Test purpose	**Call rejected after timeout TAS-CW.** User A is located in network A, user B is located in network B and subscribed to the communication Waiting service. Ensure that when user A calls user B, user A receives the 'communication Waiting indication' in the 180 Ringing provisional response if the user B is NDUB or UDUB. After timeout TAS-CW network B sends a 480 (Temporarily unavailable) response toward user A and the Reason header field is set to '19'.
Configuration	
SIP Parameter	180: Alert-Info: <urn:alert:service:call-waiting> 480: Reason: Q.850 ;cause=19
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing Timeout TAS-CW ← 480 (Temporarily unavailable) ACK →
Comments	**Check:** Is an Alert-Info header present in the 180 Ringing Response and is the value set to '<urn:alert:service:call-waiting>'? **Check:** Is a Reason header present in the 480 Response and is the protocol is set to 'Q.850' and the cause parameter set to '19'? Repeat this test in reverse direction.

Test case number	SS_cw_003
Test case group	SIP-SIP/Service/CW
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 59
Test purpose	**SIP-I support. Call Waiting indication in 180 with encapsulated ACM.** User A is located in network A, user B is located in the PSTN/PLMN part of network B and subscribed to the Call Waiting service. Ensure that when user A calls user B, an encapsulated ISUP/BICC ACM Generic notification 'call is a waiting call' is present in the 180 Ringing provisional response if the user B is NDUB.
Configuration	User B subscribed to the CW service
SIP Parameter	180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** **Backward call indicator** Called party's status indicator subscriber free **Generic notification** Notification indicator call is a waiting call
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing **Apply post test routine**
Comments	**Check:** Is an ISUP/BICC ACM present in the 180 provisional response and the Generic notification is set to 'call is a waiting cal'? Repeat this test in reverse direction.

Test case number	SS_cw_004
Test case group	SIP-SIP/Service/CW
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 59
Test purpose	**SIP-I support. Call Waiting indication in 180 with encapsulated CPG.** User A is located in network A, user B is located in the PSTN/PLMN part of network B and subscribed to the Call Waiting service. Ensure that when user A calls user B, an encapsulated ISUP/BICC CPG Generic notification 'call is a waiting call' is present in the 180 Ringing provisional response if the user B is NDUB.
Configuration	User B subscribed to the CW service
SIP Parameter	180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** **Event information** Event indicator ALERTING **Generic notification** Notification indicator call is a waiting call
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 183 Session Progress (ACM) ← 180 Ringing (CPG) **Apply post test routine**
Comments	**Check:** Is an ISUP/BICC CPG present in the 180 provisional response and the Generic notification is set to 'call is a waiting cal'? Repeat this test in reverse direction.

7.1.5.11 Explicit Communication Transfer (ECT)

Test case number	SS_ect_001
Test case group	SIP-SIP/Service/ECT
Reference	4.5.2/[11]
SELECTION EXPRESSION	[Network A] SE 37 AND [Network A] SE 11 AND [Network A] SE 49
Test purpose	**Blind/assured transfer using the REFER method.** User A is located in network A, user B and user C are located in network B. User A invokes ECT to transfer a session with user B to user C. • Ensure that a REFER request is sent from network A to network B in the dialogue with user B. The URI in the Refer-To header is set to the address of the ECT AS in network A and the method parameter is set to 'INVITE'. • Ensure that an INVITE request is sent from network B to network A and the Request URI is set to the address of the ECT AS in network A. • Ensure that an INVITE request is sent from network A to network B and the Request URI is set to the address of user C.
Configuration	
SIP Parameter	REFER: Request URI address of user B Refer-To: <URI of ECT-AS>; method=invite INVITE1 Request URI address of ECT-AS INVITE2: Request URI address of user C

Message flow

SIP (Network A)	**Interconnection Interface**	**SIP (Network B)**
	A confirmed session is established between user A and user B	
	A confirmed session is established between user A and user C	
	User A invokes ECT to transfer the session to user C	
	REFER	→
←	202 Accepted	
←	NOTIFY (100)	
	200 OK NOTIFY	→
	CASE Blind transfer	
	BYE (A-B)	→
←	200 OK BYE	
←	INVITE1 (ECT-AS)	
	INVITE2 (user C)	→
←	200 OK INVITE	
	ACK	→
	200 OK INVITE	→
←	ACK	
←	NOTIFY (200)	
	200 OK NOTIFY	→
	CASE Assured transfer	
	BYE (A-B)	→
←	200 OK BYE	
	Apply post test routine	

Comments	**Check:** Is a REFER request is sent network B, the Refer-To header is set to the URI of the ECT-AS in network A and a method parameter is present set to 'INVITE'? **Check:** Is a NOTIFY request sent to network A containing sipfrag body set to 'SIP/2.0 100 Trying' and if Blind transfer is applicable the session from user A to user B is terminated by user A? **Check:** Is an INVITE request sent to network A the Request line is set to the address of the ECT-AS in network A? **Check:** Is an INVITE request is sent to network B the Request is set to the address of user C? **Check:** When the session from user B to user C is confirmed a NOTIFY request is sent to network A containing sipfrag body set to 'SIP/2.0 200 OK' and if Assured transfer is applicable the session from user A to user B is terminated by user A? **Check:** Ensure the property of speech between user B and user C. Repeat this test in reverse direction.

Test case number	SS_ect_002
Test case group	SIP-SIP/Service/ECT
Reference	4.5.2/[11]
SELECTION EXPRESSION	[Network A] SE37 AND [Network A] SE 11 AND [Network A] SE 50
Test purpose	**Consultative transfer using the REFER method.** User A is located in network A, user B and user C are located in network B. User A invokes ECT to transfer a session with user B to user C. • Ensure that a REFER request is sent from network A to network B in the dialogue with user B. The URI in the Refer-To header is set to the address of the ECT AS in network A and the method parameter is set to 'INVITE'. • Ensure that an INVITE request is sent from network B to network A and the Request URI is set to the address of the ECT AS in network A. • Ensure that an INVITE request is sent from network A to network B and the Request URI is set to the address of user C and a Replaces header is present containing the session identifiers of the session A - C.
Configuration	
SIP Parameter	REFER:Request URI address of user B Refer-To: <URI of ECT-AS>; method=invite INVITE1 Request URI address of ECT-AS INVITE2: Request URI address of user C Require: replaces Replaces: <session A-C>

Message flow

SIP (Network A)	**Interconnection Interface**	**SIP (Network B)**
	A confirmed session is established between user A and user B	
	A confirmed session is established between user A and user C	
	User A invokes ECT to transfer the session to user C	
	REFER →	
	← 202 Accepted	
	← NOTIFY (100)	
	200 OK NOTIFY →	
	← INVITE1 (ECT-AS)	
	INVITE2 (user C) →	
	← 200 OK INVITE	
	ACK →	
	200 OK INVITE →	
	← ACK	
	← NOTIFY (200)	
	200 OK NOTIFY →	
	BYE (A-B) →	
	← 200 OK BYE	
	← BYE (A-C)	
	200 OK BYE →	
	Apply post test routine	

Comments	**Check:** Is a REFER request is sent network B, the Refer-To header is set to the URI of the ECT-AS in network A and a method parameter is present set to 'INVITE'? **Check:** Is an INVITE request sent to network A the Request line is set to the address of the ECT-AS in network A? **Check:** Is an INVITE request is sent to network B the Request is set to the address of user C and a Replaces header is present contains the session identifiers of the session A-C? **Check:** Is the session A - B and the session A - C terminated? **Check:** Ensure the property of speech between user B and user C. Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_ect_003</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/ECT</td></tr>
<tr><td>Reference</td><td>4.5.2/[11], 4.7.2.9.7/[20]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE37 AND NOT [Network A] SE 12 AND [Network A] SE 49</td></tr>
<tr><td>Test purpose</td><td>Blind/assured transfer using the 3pcc method.

User A is located in network A, user B an user C are located in network B User A invokes ECT to transfer a session with user B to user C.
• Ensure that the network A establishes a session to user C.
• Ensure that the network A sends a reINVITE to update the session between user A and user B (SDP: IP address, port and codec).</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td>INVITE1 Request URI address of user C

INVITE2: Request URI address of user B
SDP
c=IN IP4/6 [new IP address]
m=audio [new port] RTP/AVP [new codec list]
a=[new attributes]</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
A confirmed session is established between user A and user B
User A invokes ECT to transfer the session to user C
INVITE1 (user C) →
← 180 Ringing
← 200 OK INVITE
ACK →

INVITE2 (user B) →
← 200 OK INVITE
ACK →
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Check: Is an initial INVITE is sent from network A to user C to establish a dialogue between network A and user C?
Check: Is a reINVITE is sent from network A to user B update the session parameter in the SDP?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_ect_004
Test case group	SIP-SIP/Service/ECT
Reference	4.5.2/[11], 4.7.2.9.7/[20]
SELECTION EXPRESSION	[Network A] SE37 AND [Network A] SE 12 AND [Network A] SE 50
Test purpose	**Consultative transfer using the 3pcc method.** User A is located in network A, user B and user C are located in network B User A invokes ECT to transfer a session with user B to user C. • Ensure that the network A sends a reINVITE to update the session between user A and user B (SDP: IP address, port and codec). • Ensure that the network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec).
Configuration	
SIP Parameter	INVITE1: Request URI address of user C SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes]
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **A confirmed session is established between user A and user B** **A confirmed session is established between user A and user C** **User A invokes ECT to transfer the session to user C** INVITE1 (user B) → ← 200 OK INVITE ACK → INVITE2 (user C) → ← 200 OK INVITE ACK → **Apply post test routine**	
Comments	**Check:** Is a reINVITE is sent from network A to user B update the session parameter in the SDP. **Check:** Is a reINVITE is sent from network A to user C update the session parameter in the SDP. Repeat this test in reverse direction.

Test case number	SS_ect_005
Test case group	SIP-SIP/Service/ECT
Reference	5.4.3.2/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 60
Test purpose	**SIP-I support. Call Transfer invoked in active state, call was previous on HOLD.** BICC/ISUP - SIP-I interworking applies in the originating network User A and C are located in network A and user B is located in network B. Ensure that an User A can successfully invoke the ECT supplementary service and transfer the call with User B to User C in active state.
Configuration	User A is subscribed to the Explicit Call Transfer supplementary service
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/sdp a=sendrecv --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **FAC** Generic Notification Call transfer active Call transfer number --[any boundary name]--
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **A confirmed session is established between user A and user B and set on hold** **User A invokes ECT to transfer the session to user C** *INFO (LOP request)* → ← *200 OK INFO* ← *INFO (LOP response)* *200 OK INFO* → **CASE A** INVITE (sendrecv; **FAC**) → ← 200 OK INVITE ACK → **CASE B** INFO (**FAC**) → ← 200 OK INFO INVITE (sendrecv) → ← 200 OK INVITE ACK → **Apply post test routine**	
Comments	A session from User A to User B is already established User A sets the User B on hold User A invokes the ECT service **Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**request**'? **Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**response**'? **Check:** Is (CASE A) an INVITE request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer active' and in addition the media stream is set to 'sendrecv'? **Check:** Is (CASE B) an INFO request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer active'? In addition is an **INVITE** request sent and the media stream is set to 'sendrecv' to resume the held session? **NOTE:** The content of the FAC in the INVITE request is Equal to the content of the FAC in the INFO request. Repeat this test in reverse direction.

Test case number	SS_ect_006
Test case group	SIP-SIP/Service/ECT
Reference	5.4.3.2/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 60
Test purpose	**SIP-I support. Call Transfer invoked in alerting state, call was previous on HOLD.** BICC/ISUP - SIP-I interworking applies in the originating network User A and C are located in network A and user B is located in network B. Ensure that an User A can successfully invoke the ECT supplementary service and transfer the call with User B to User C in alerting state.
Configuration	User A is subscribed to the Explicit Call Transfer supplementary service
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/sdp a=sendrecv --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **FAC** Generic Notification Call transfer alerting Call transfer number --[any boundary name]--
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **A confirmed session is established between user A and user B and set on hold** **User A invokes ECT to transfer the session to user C** *INFO (LOP request)* → ← *200 OK INFO* ← *INFO (LOP response)* *200 OK INFO* → **CASE A** INVITE (sendrecv; **FAC**) → ← 200 OK INVITE ACK → **CASE B** INFO (**FAC**) → ← 200 OK INFO INVITE (sendrecv) → ← 200 OK INVITE ACK → **Apply post test routine**	
Comments	A session from User A to User B is already established User A sets the User B on hold A session from User A to User C is already established User A invokes the ECT service **Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**request**'? **Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**response**'? **Check:** Is (CASE A) an INVITE request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer alerting' and in addition the media stream is set to 'sendrecv'? **Check:** Is (CASE B) an INFO request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer alerting'? In addition is an **INVITE** request sent and the media stream is set to 'sendrecv' to resume the held session? **NOTE:** The content of the FAC in the INVITE request is Equal to the content of the FAC in the INFO request. Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_ect_007</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/ECT</td></tr>
<tr><td>Reference</td><td>5.4.3.2/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 17 AND SE 47 AND SE 60</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. Call Transfer invoked in active state.

BICC/ISUP - SIP-I interworking applies in the originating network Users A and B are located in network A and User C is located in network B.
Ensure that an User A can successfully invoke the ECT supplementary service and transfer the call with User B to User C in active state.</td></tr>
<tr><td>Configuration</td><td>User A is subscribed to the Explicit Call Transfer supplementary service</td></tr>
<tr><td>SIP Parameter</td><td>INFO
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
FAC
Generic Notification
Call transfer active
Call transfer number</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
A confirmed session is established between user A and user C
User A invokes ECT to transfer the session to user C
INFO (LOP request) →
← 200 OK INFO
← INFO (LOP response)
200 OK INFO →

INFO (FAC) →
← 200 OK INFO
Apply post test routine</td></tr>
<tr><td>Comments</td><td>A session from User A to User B is already established
User A sets the User B on hold
A session from User A to User C is already established
User A invokes the ECT service
Check: Is (optional) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to 'request'?
Check: Is (optional) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to 'response'?
Check: Is (CASE B) an INFO request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer active'?
NOTE: The content of the FAC in the INVITE request is Equal to the content of the FAC in the INFO request.
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_ect_008
Test case group	SIP-SIP/Service/ECT
Reference	5.4.3.2/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 60
Test purpose	**SIP-I support. Call Transfer invoked in alerting state.** BICC/ISUP - SIP-I interworking applies in the originating network User A and B are located in network A and user C is located in network B. Ensure that an User A can successfully invoke the ECT supplementary service and transfer the call with User B to User C in alerting state.
Configuration	User A is subscribed to the Explicit Call Transfer supplementary service
SIP Parameter	INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **CPG** Generic Notification Call transfer alerting Call transfer number
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **A session in the early dialogue is established between user A and user C** **User A invokes ECT to transfer the session to user C** *INFO (LOP request)* → ← *200 OK INFO* ← *INFO (LOP response)* *200 OK INFO* → INFO (**CPG**) → ← 200 OK INFO **Apply post test routine**	
Comments	A session from User A to User B is already established User A sets the User B on hold A session from User A to User C is already established User A invokes the ECT service **Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**request**'? **Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**response**'? **Check:** Is (CASE B) an INFO request sent and an ISUP CPG message is present containing a Generic notification indicator is set to 'Call transfer alerting'? **NOTE:** The content of the FAC in the INVITE request is Equal to the content of the FAC in the INFO request. Repeat this test in reverse direction.

7.1.5.12 Malicious Communication Identification (MCID)

Test case number	SS_mcid_001
Test case group	SIP-SIP/Service/MCID
Reference	4.5.2.5/[18]
SELECTION EXPRESSION	SE 38
Test purpose	**Network B sends a MCID request, no response.** User A is located in network A, user B is located in network B and subscribed to the Malicious Communication Identification service. When user A call user B and no originating identification is present in the INVITE request, the network B sends an INFO request to network A requesting the originating identity. After timeout of timer TO-ID the network B sends the 180 Ringing response.
Configuration	User B is subscribed to the MCID service
SIP Parameter	INFO: <…:mcid……………....> <…:request> <…:McidRequestIndicator>01</…:McidRequestIndicator> <…:HoldingIndicator >…</…:HoldingIndicator> </…:request> </…:mcid>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← INFO 200 OK INFO → **Timeout T_{O-ID}** ← 180 Ringing **Apply post test routine**
Comments	**Check:** Is an INFO request sent to network A? **Check:** Is the McidRequestIndicator element set to ,01'? **Check:** is a 200 OK INFO response sent to network B? Repeat this test in reverse direction.

Test case number	SS_mcid_002
Test case group	SIP-SIP/Service/MCID
Reference	4.5.2.5/[18]
SELECTION EXPRESSION	SE 38 AND SE 47
Test purpose	**Network B sends a MCID request, MCID response.** PSTN user A is located in network A, user B is located in network B and subscribed to the Malicious Communication Identification service. When user A call user B and no originating identification is present in the INVITE request, the network B sends an INFO request to network B requesting the originating identity. After receipt of an INFO request from network A the network B sends the 180 Ringing response.
Configuration	User B subscribed to the MCID service User A is a ISDN or POTS user in the PSTN of network A
SIP Parameter	INFO: <…:mcid …………………....> <…:request> <…:McidRequestIndicator>01</…:McidRequestIndicator> <…:HoldingIndicator >…</…:HoldingIndicator> </…:request> </…:mcid> INFO: <…:mcid………………......> <…:response> <…:McidResponseIndicator>01</…:McidResponseIndicator> <…:HoldingProvidedIndicator>…</…:HoldingProvidedIndicator> <…:OrigPartyIdentity>any URI</…:OrigPartyIdentity> <…:OrigPartyPresentationRestriction> true/false </…:OrigPartyPresentationRestriction> </…:response> </…:mcid>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← INFO 200 OK INFO → INFO → ← 200 OK INFO ← 180 Ringing **Apply post test routine**
Comments	**Check:** Is an INFO request sent to network A? **Check:** Is the McidRequestIndicator element set to ,01'? **Check:** Is a 200 OK INFO response sent to network B? **Check:** Is an INFO request sent to network B? **Check:** Is the McidResponseIndicator element set to ,01'? **Check:** Is the OrigPartyIdentity element present in the response element? **Check:** Is a 200 OK INFO response sent to network A? **A INFO request containing a mcid response element sent by the MGCF in network A is optional.** Repeat this test in reverse direction.

Test case number	SS_mcid_003
Test case group	SIP-SIP/Service/MCID
Reference	5.4.3.2/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 61
Test purpose	**SIP-I support. Network B sends a MCID request, no response.** User A is located in network A, user B is located in the PSTN/PLMN part of network B and subscribed to the Malicious Call Identification service. When user A call user B and no originating identification is present in the INVITE request, the network B sends an INFO request to network A and an ISUP/BICC IDR message is present the MCID request indicator is set to 'MCID requested' requesting the originating identity. After timeout of timer (ISUP) T39 the network B sends the 180 Ringing response.
Configuration	User B is subscribed to the MCID service
SIP Parameter	INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IDR** **MCID request indicators** MCID request indicator MCID requested
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← INFO(IDR) 200 OK INFO → **Timeout T_{O-ID}** ← 180 Ringing **Apply post test routine**
Comments	**Check:** Is an INFO request sent to network A? **Check:** Is a ISUP/BICC IDR message is present and the MCID request indicator is set to 'MCID requested'? **Check:** Is a 200 OK INFO response sent to network B? **NOTE:** Based on network policies the MCID request indicator can be set to'MCID not requested'. Repeat this test in reverse direction.

Test case number	SS_mcid_004
Test case group	SIP-SIP/Service/MCID
Reference	5.4.3.2/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 61
Test purpose	**SIP-I support. Network B sends a MCID request, MCID response.** PSTN user A is located in network A, user B is located in the PSTN/PLMN part of network B and SIP-I - ISUP/BICC interworking applies and User B is subscribed to the Malicious Call Identification service. When user A call user B and no originating identification is present in the INVITE request, the network B sends an INFO request to network B requesting the originating identity. After receipt of an INFO request from network A the network B sends the 180 Ringing response.
Configuration	User B subscribed to the MCID service User A is a ISDN or POTS user in the PSTN of network A
SIP Parameter	INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IDR** **MCID request indicators** MCID request indicator MCID requested INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IRS** **MCID response indicators** MCID response indicator MCID included **Calling party number**
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← INFO(IDR) 200 OK INFO → INFO(IRS) → ← 200 OK INFO ← 180 Ringing **Apply post test routine**
Comments	**Check:** Is an INFO request sent to network A and a ISUP/BICC IDR is present and the MCID request indicator is set to 'MCID requested'? **Check:** Is a 200 OK INFO response sent to network B? **Check:** Is an INFO request sent to network B and a ISUP/BICC IRS is present and the MCID response indicator is set to 'MCID included'? **Check:** Is the Calling party number present in the attached ISUP/BICC IRS? **Check:** Is a 200 OK INFO response sent to network A? Repeat this test in reverse direction.

7.1.5.13 Message Waiting Indication (MWI)

Test case number	SS_mwi_001
Test case group	SIP-SIP/Service/MWI
Reference	4.7.2/[16]
SELECTION EXPRESSION	[Network A] SE 39 AND [Network B] SE 39
Test purpose	**Initial subscription of a Voicemail box.** The Voicemail owner is in network A, his Voicemail box is located in network B. Ensure that a Voicemail owner is able to activate his Voicemail box.
Configuration	Voicemail in network B Voicemail owner in network A
SIP Parameter	SUBCRIBE Event: message-summary Expires: [any value] Accept: application/simple-message-summary NOTIFY Subscription-State: active;expires=[any value] Event: message-summary
Message flow	
SIP (Network A)	**Interconnection Interface** / **SIP (Network B)**
	SUBCRIBE →
	← 200 OK SUBSCRIBE
	← NOTIFY
	200 OK NOTIFY →
	← 200 OK BYE
	← NOTIFY
	200 OK NOTIFY →
	Apply post test routine
Comments	**Check:** Is it possible for a user in network A to subscribe to a Voicemail box in network B? **Check:** Is the Event header in the SUBCRIBE set to 'message-summary'? **Check:** Is the Accept header in the SUBCRIBE set to 'application/simple-message-summary'? **Check:** Is the Event header in the NOTIFY is set to 'message-summary'? Repeat this test in reverse direction.

Test case number	SS_mwi_002
Test case group	SIP-SIP/Service/MWI
Reference	4.7.2/[16]
SELECTION EXPRESSION	[Network A] SE 39 AND [Network B] SE 39
Test purpose	**A new entry in the Voicemail box is indicated to the owner.** The Voicemail owner is in network A, his Voicemail box is located in network B. Ensure when a user calls user A and the call is not answered, the call is forwarded to the Voicemail box of user A in network B. Ensure that the user A is notified by message waiting indication that there is a new message present in his voicemail account.
Configuration	Voicemail in network B Voicemail owner in network A
SIP Parameter	NOTIFY Subscription-State: active;expires=[any value] Event: message-summary Content-Type: application/simple-message-summary Messages-Waiting: yes Message-Account: sip:userA@networkA (optional) Voice-Message: [any new value]/[any old value] (optional)

Message flow

SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE	→	
	←	200 OK INVITE		
		ACK	→	
		BYE	→	
	←	200 OK BYE		
	←	NOTIFY		
		200 OK NOTIFY	→	

Apply post test routine

Comments	**Check:** Is the Event header in the NOTIFY set to 'message-summary'? **Check:** Is the Content-Type header in the NOTIFY set to 'application/simple-message-summary'? **Check:** Contains the MIME body the header 'Messages-Waiting' set to 'yes'? **Check:** Contains the MIME body the optional header 'Message-Account'? **Check:** Contains the MIME body the optional header 'Voice-Message'? Repeat this test in reverse direction.

7.1.5.14 Completion of Communications to Busy Subscriber (CCBS), Completion of Communications by No Reply (CCNR)

Test case number	SS_cc_001
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.3/[14]
SELECTION EXPRESSION	[Network A] SE 40 AND [Network B] SE 40
Test purpose	**Indicating of CCBS possible.** User A is located in network A and user B is located in network B. Ensure when user A calls user B and user B is busy, the network B sends an indication that CCBS is possible in the 486 Busy Here final response.
Configuration	
SIP Parameter	486: Call-Info: <sip:UE-B>;purpose=call-completion;m=BS
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 486 Busy Here ACK →
Comments	**Check:** The 486 final response contains the Call-Info header. **Check:** The Call-Info header contains the URI of user B as the monitor point in network B. **Check:** The Call-Info header contains the purpose parameter set to 'call-completion' and the m parameter set to 'BS'. Repeat this test in reverse direction.

Test case number	SS_cc_002
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.3/[14]
SELECTION EXPRESSION	[Network A] SE 41 AND [Network B] SE 41
Test purpose	**Indicating of CCNR possible.** User A is located in network A and user B is located in network B. Ensure when user A calls user B and user B is free, the network B sends an indication that CCNR is possible in the 180 Ringing provisional response.
Configuration	
SIP Parameter	180: Call-Info: <sip:UE-B>;purpose=call-completion;m=NR
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing **Apply post test routine**
Comments	**Check:** The 180 provisional response contains the Call-Info header. **Check:** The Call-Info header contains the URI of user B as the monitor point in network B. **Check:** The Call-Info header contains the purpose parameter set to 'call-completion' and the m parameter set to 'NR'. Repeat this test in reverse direction.

Test case number	SS_cc_003
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.2/[14]
SELECTION EXPRESSION	([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR [Network B] SE 41)
Test purpose	**Invocation of CCBS or CCNR.** User A is located in network A and user B is located in network B. • Ensure when user A call user B and user B is busy, the indication that CCBS is possible is sent to the network A. when user A invokes CCBS, a SUBSCRIBE request is sent to the network B, the Event header is set to 'call-completion' and the m parameter in the Request line is set to 'BS'. • Ensure when user A call user B and user B is free, the indication that CCNR is possible is sent to the network A. when user A invokes CCNR, a SUBSCRIBE request is sent to the network B, the Event header is set to 'call-completion' and the m parameter in the Request line is set to 'NR'. Ensure that the network B sends a NOTIFY request to network A to confirm that the request is in the Call completion queue at the terminating Application Server.
Configuration	
SIP Parameter	SUBSRIBE sip:B-AS;m=**BS** or m=**NR** From:<UE-A> To:<UE-B> Contact:<A-AS> Event:call-completion NOTIFY sip:A-AS Event:call-completion Content-Type: application/call-completion state: queued service-retention
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **An indication whether CCBS or CCNR is possible is sent by network B** SUBSCRIBE → ← 202 Accepted ← NOTIFY 200 OK NOTIFY → **Apply post test routine**	
Comments	**Check:** Is a SUBCRIBE request is sent to network B? **Check:** Is the m parameter in the Request URI is set to '**BS'** in case of CCBS request or set to '**NR'** in case of CCNR? **Check:** Is a NOTIFY request is sent to network A and the Event header is set to 'call-completion' and the state header in the message body is set to 'queued''. Repeat this test in reverse direction. NOTE: The service-retention header in the NOTIFY body is a network option.

Test case number	SS_cc_004
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.3/[14]
SELECTION EXPRESSION	([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR [Network B] SE 41)
Test purpose	**Invocation of CCBS or CCNR unsuccessful; short term denial** User A is located in network A and user B is located in network B. Ensure that user A invokes a CCBS or CCNR request to network B and the network B is currently unable to process the request (e.g. the B-queue is full), a 480 Temporarily Unavailable final response is sent.
Configuration	
SIP Parameter	SUBSRIBE sip:B-AS;m=BS or m=NR From:<UE-A> To:<UE-B> Contact:<A-AS> Event:call-completion
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **An indication whether CCBS or CCNR is possible is sent by network B** SUBSCRIBE → ← 480 (Temporarily Unavailable)	
Comments	**Check:** Is a SUBCRIBE request is sent to network B? **Check:** Is the m parameter in the Request URI is set to 'BS' in case of CCBS request or set to 'NR' in case of CCNR? **Check:** Is a 480 Temporarily Unavailable sent from network B indicates the CCBS or CCNR request is unsuccessful e.g. CC queue is full? Repeat this test in reverse direction.

Test case number	SS_cc_005
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.3/[14]
SELECTION EXPRESSION	([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR [Network B] SE 41)
Test purpose	**Successful CC operation** User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS or CCNR request. • Ensure when the user B becomes available for CC recall, the CC recall procedure is started. The network B sends a NOTIFY request to network A and a state header is present in the message body set to 'ready'. • Ensure that the recall from user A to user B is successful. • Ensure that a CC revocation notification is dent to network A to indicate the subscription is terminated; the reason header is set to 'noresource'.
Configuration	
SIP Parameter	NOTIFY sip:O-AS Event:call-completion Content-Type: application/call-completion state: ready NOTIFY sip:O-AS Event:call-completion Subscription-State: terminated; reason=noresource
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		A CCBS or CCNR request was already successful		
	←	NOTIFY		
		200 OK NOTIFY	→	
		INVITE	→	
	←	180 Ringing		
	←	NOTIFY		
		200 OK NOTIFY	→	
	←	200 OK INVITE		
		ACK	→	
		Apply post test routine		

Comments	**Check:** Is a NOTIFY request is sent to network A and the Event header is set to 'call-completion' and the state header in the message body is set to 'ready'? **Check:** Is the recall from user A to user B is successful? **Check:** Is the CC revocation is performed after the 180 Ringing or the 200 OK INVITE was sent to user A? Repeat this test in reverse direction.

Test case number	SS_cc_006
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.31/[14]
SELECTION EXPRESSION	([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR [Network B] SE 41)
Test purpose	**No CC call as result.** User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS or CCNR request. Ensure when no recall result is performed while CC-T9 is running (user A does not calls to user B) the network B sends a NOTIFY request to network A with an indication that the subscription is terminated, the reason header is set to 'rejected'.
Configuration	
SIP Parameter	NOTIFY sip:O-AS Event:call-completion Content-Type: application/call-completion state: ready NOTIFY sip:O-AS Event:call-completion Subscription-State: terminated; reason=rejected
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** **A CCBS or CCNR request was already successful** **User B is available for recall** ← NOTIFY 200 OK NOTIFY → **CC-T9 expires** ← NOTIFY 200 OK NOTIFY →	
Comments	**Check:** Is a NOTIFY request is sent to network A and the Event header is set to 'call-completion' and the state header in the message body is set to 'ready'? User A does not perform the recall **Check:** Is the CC revocation is performed after timer CC-T9 expires? Repeat this test in reverse direction.

Test case number	SS_cc_007
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.2/[14]
SELECTION EXPRESSION	([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR [Network B] SE 41)
Test purpose	**User A is unavailable while CC recall is performed.** User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS or CCNR request. User B is available for CC-recall and network B sends a CC-recall notification to network A. • Ensure that network A sends PUBLISH request to suspend the recall procedure. • Ensure that network A sends PUBLISH request to resume the recall procedure if user A is available to complete the recall procedure. • Ensure the network B sends a NOTIFY request to indicate the CC-recall procedure.
Configuration	
SIP Parameter	NOTIFY sip:O-AS Event:call-completion Content-Type: application/call-completion state: ready PUBLISH sip B-AS To: SIP 2 Event: presence Content-Type: application/pidf+xml <?xml version="1.0" encoding="UTF-8"?> <presence <status> <basic>closed</basic> PUBLISH sip B-AS To: SIP 2 Event: presence Content-Type: application/pidf+xml <?xml version="1.0" encoding="UTF-8"?> <presence <status> <basic>open</basic>
Message flow	
Comments	

SIP (Network A)	**Interconnection Interface**	**SIP (Network B)**
	A CCBS or CCNR request was already successful	
	User B is available for recall	
←	NOTIFY	
	200 OK NOTIFY →	
	User A is busy	
	PUBLISH →	
←	200 OK PUBLISH	
	User A is no longer busy	
	PUBLISH →	
←	200 OK PUBLISH	
	User B is available for recall	
←	NOTIFY	
	200 OK NOTIFY →	
	Apply post test routine	

Test case number	SS_cc_008
Test case group	SIP-SIP/Service/CC
Reference	6.11.2/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	**SIP-I support: Indicating of CCBS possible.** BICC/ISUP - SIP-I interworking applies in the terminating network and User A is located in network A and user B is located in network B. Ensure when user A calls user B and user B is busy, the network B sends a 486 Busy Here final response and an encapsulated ISUP REL is present, the Cause value indicator is set to #17 or #34 and the CCBS possible indicator is set to 'CCBS possible'.
Configuration	
SIP Parameter	486: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **REL** Cause value #17 or #34 Diagnostics CCBS possible
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 486 Busy Here (**REL**) ACK →
Comments	**Check:** The 486 final response contains an encapsulated BICC/ISUP REL, the Cause value set to 17 or 34 and the Diagnostics set to 'CCBS possible'. Repeat this test in reverse direction.

Test case number	SS_cc_009
Test case group	SIP-SIP/Service/CC
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	**SIP-I support: Indicating of CCNR possible.** BICC/ISUP - SIP-I interworking applies in the terminating network User A is located in network A and user B is located in network B. Ensure when user A calls user B and user B is free, the network B sends a 180 Ringing provisional response and an encapsulated ACM is present containing a CCNR possible indicator set to 'CCNR possible'.
Configuration	
SIP Parameter	180: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** CCNR possible indicator CCNR possible
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing (**ACM**) **Apply post test routine**
Comments	**Check:** The 180 provisional response contains an encapsulated ACM. **Check:** The CCNR possible indicator in the ACM is set to 'CCNR possible'. Repeat this test in reverse direction.

7.1.6 Other PSTN services (SIP-I interworking)

7.1.6.1 User-to-User Signaling (UUS)

Test case number	SS_uus_001
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 1 implicit in initial INVITE request.** BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 1 implicit request calls user B an **User-to-user Information** parameter is present in the encapsulated IAM of the initial INVTE request.
Configuration	User A is subscribed to the User-to-User service 1 implicit request
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Information User Information
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE (IAM) ➔ **Apply post test routine**
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request? **Check:** Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM? Repeat this test in reverse direction.

Test case number	SS_uus_002
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1, 6.5/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 1 implicit response in 180.** BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 1 implicit request calls user B subscribed to User-to-User service 1 an **User-to-user Information** parameter is present in the encapsulated ACM of the 180 response.
Configuration	User A is subscribed to the User-to-User service 1 implicit request
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Information User Information 180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** User-to-user Information User Information
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE (IAM) → ← 180 Ringing (ACM) **Apply post test routine**
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request? **Check:** Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM? **Check:** Is an ISUP/BICC ACM encapsulated in the 180 response? **Check:** Is a User-to-user Information parameter present in the encapsulated ISUP/BICC ACM? Repeat this test in reverse direction.

Test case number	SS_uus_003
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 1 explicit in initial INVITE request.** BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 1 explicit request calls user B an **User-to-user Indicator** parameter is present set to 'Request service 1', 'not essential' or 'essential' in the encapsulated IAM of the initial INVTE request.
Configuration	User A is subscribed to the User-to-User service 1 explicit request
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Indicator Request service 1 not essential or essential User-to-user Information User Information
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE (IAM) ➔ **Apply post test routine**	
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request? **Check:** Is a User-to-user Indicator parameter present in the encapsulated ISUP/BICC IAM? **Check:** Is the **Request service 1** set to 'not essential' or 'essential'? Repeat this test in reverse direction.

Test case number	SS_uus_004
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1, 6.5/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 1 explicit response in 180.** BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 1 explicit request calls user B subscribed to User-to-User service 1 an **User-to-user Indicator** parameter is present set to 'Response', 'service 1 provided' in the encapsulated ACM of the 180 response.
Configuration	User A is subscribed to the User-to-User service 1 explicit request
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Indicator Request service 1 essential or not essential 180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** User-to-user Indicator Response service 1 provided
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE (IAM) → ← 180 Ringing (ACM) **Apply post test routine**	
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request? **Check:** Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM? **Check:** Is an ISUP/BICC ACM encapsulated in the 180 response? **Check:** Is an User-to-user Indicator parameter present set to 'Response', 'service 1 provided' in the encapsulated ISUP/BICC ACM? Repeat this test in reverse direction.

Test case number	SS_uus_005
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1, 6.5/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 1 not essential explicit rejected in 180.** BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 1 explicit request calls user B not subscribed to User-to-User service 1 the call is rejected by the network an **User-to-user Indicator** parameter is present set to 'Response', 'service 1 not provided' in the encapsulated ACM of the 180 response.
Configuration	User A is subscribed to the User-to-User service 1 explicit request User B is not subscribed to the User-to-User service 1
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Indicator Request service 1 not essential 180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** User-to-user Indicator Response service 1 not provided
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE (IAM) → ← 180 Ringing (ACM) **Apply post test routine**
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request? **Check:** Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM? **Check:** Is an ISUP/BICC ACM encapsulated in the 180 response? **Check:** Is an User-to-user Indicator parameter present set to 'Response', 'service 1 not provided' in the encapsulated ISUP/BICC ACM? Repeat this test in reverse direction.

Test case number	SS_uus_006
Test case group	SIP-SIP/SIP-I/UUS
Reference	6.11.2, 7.1/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 1 essential explicit rejection.** BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 1 explicit request calls user B subscribed to User-to-User service 1 essential is rejected by the network or by the user. A 500 Server Internal Error is sent and an encapsulated ISUP/BICC REL is present, the Cause value is set to #29 or #69.
Configuration	User A is subscribed to the User-to-User service 1 explicit request
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Indicator Request service 1 essential 500 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **REL** Cause value #29 or #69
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE (IAM) → ← 500 Server Internal Error (REL) ACK → **Apply post test routine**
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request? **Check:** Is a User-to-user Indicator parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 1', 'essential'? **Check:** Is an ISUP/BICC REL encapsulated in the 500 response? **Check:** Is the Cause value set to #29 or #69 in the encapsulated REL? Repeat this test in reverse direction.

Test case number	SS_uus_007
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 2 in initial INVITE request.** BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 2 calls user B an **User-to-user Indicator** parameter is present set to 'Request service 2', 'not essential' or 'essential' in the encapsulated IAM of the initial INVTE request.
Configuration	User A is subscribed to the User-to-User service 2
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Indicator Request service 2 not essential or 'essential'
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE (IAM) ➔ **Apply post test routine**	
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request and the a User-to-user Indicator parameter is set to Is the **Request service 2** 'not essential' or 'essential'? Repeat this test in reverse direction.

Test case number	SS_uus_008
Test case group	SIP-SIP/SIP-I/UUS
Reference	5.4.3.2, 6.5, 7.1/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 2 in initial INVITE request successful.** BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 2 calls user B an **User-to-user Indicator** parameter is present set to 'Request service 2', 'not essential' or 'essential' in the encapsulated IAM of the initial INVTE request. The User-to-User service is successful.
Configuration	User A is subscribed to the User-to-User service 2 User B is subscribed to the User-to-User service 2
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Indicator Request service 2 not essential or 'essential' 180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** User-to-user Indicator Response service 2 provided INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **USR** User-to-user Information User Information 183 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **USR** User-to-user Information User Information
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE (IAM) → ← 180 Ringing (ACM) INFO (USR) → ← 200 OK INFO ← 183 Session Progress (USR) **Apply post test routine**	
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request and the a User-to-user Indicator parameter is set to Is the **Request service 2** 'not essential' or 'essential'? **Check:** Is an ISUP/BICC ACM encapsulated in the 180 and the User-to-user Indicator parameter is set to 'Response', 'service 2 provided'? **Check:** Is an ISUP/BICC USR encapsulated in the INFO message sent from network A to network B containing an User-to-user Information parameter? **Check:** Is an ISUP/BICC USR encapsulated in the 183 response sent from network B to network A containing an User-to-user Information parameter? Repeat this test in reverse direction.

Test case number	SS_uus_009
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1, 6.5/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 2 not essential rejected in 180 response.** BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 2 not essential calls user B not subscribed to User-to-User service 2 the call is rejected by the network an **User-to-user Indicator** parameter is present set to 'Response', 'service 2 not provided' in the encapsulated ACM of the 180 response.
Configuration	User A is subscribed to the User-to-User service 2 User B is not subscribed to the User-to-User service 2
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Indicator Request service 2 not essential 180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ACM** User-to-user Indicator Response service 2 not provided
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE (IAM) → ← 180 Ringing (ACM) **Apply post test routine**	
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request? **Check:** Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 2' 'not essential'? **Check:** Is an ISUP/BICC ACM encapsulated in the 180 response? **Check:** Is an User-to-user Indicator parameter present set to 'Response', 'service 2 not provided' in the encapsulated ISUP/BICC ACM? Repeat this test in reverse direction.

Test case number	SS_uus_010
Test case group	SIP-SIP/SIP-I/UUS
Reference	6.11.2, 7.1/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 2 essential rejection.** BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 2 essential calls user B not subscribed to User-to-User service 2 the call is rejected by the network. A 500 Server Internal Error is sent and an encapsulated ISUP/BICC REL is present, the Cause value is set to #29 or #69.
Configuration	User A is subscribed to the User-to-User service 2
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Indicator Request service 2 essential 500 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **REL** Cause value #29 or #69
Message flow	**SIP (Network A)** — **Interconnection Interface** — **SIP (Network B)** INVITE (IAM) → ← 500 Server Internal Error (REL) ACK → **Apply post test routine**
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request? **Check:** Is a User-to-user Indicator parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 1', 'essential'? **Check:** Is an ISUP/BICC REL encapsulated in the 500 response? **Check:** Is the Cause value set to #29 or #69 in the encapsulated REL? Repeat this test in reverse direction.

Test case number	SS_uus_011
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 3 in initial INVITE request.** BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 3 calls user B an **User-to-user Indicator** parameter is present set to 'Request service 3', 'not essential' or 'essential' in the encapsulated IAM of the initial INVTE request.
Configuration	User A is subscribed to the User-to-User service 3
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Indicator Request service 3 not essential or 'essential'
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE (IAM) ➔ **Apply post test routine**
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request and the a User-to-user Indicator parameter is set to Is the **Request service 3** 'not essential' or 'essential'? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_uus_012</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/SIP-I/UUS</td></tr>
<tr><td>Reference</td><td>5.4.3.2, 6.5, 7.1/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63</td></tr>
<tr><td>Test purpose</td><td>SIP-I support: Indicating of User-to-User service 3 in initial INVITE request successful.

BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B.
Ensure when user A subscribed to the User-to-User service 3 calls user B an User-to-user Indicator parameter is present set to 'Request service 3', 'not essential' or 'essential' in the encapsulated IAM of the initial INVTE request. The User-to-User service is successful.</td></tr>
<tr><td>Configuration</td><td>User A is subscribed to the User-to-User service 3
User B is subscribed to the User-to-User service 3</td></tr>
<tr><td>SIP Parameter</td><td>INVITE:
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
IAM
User-to-user Indicator
Request
service 3
not essential or 'essential'

200 OK
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
ANM
User-to-user Indicator
Response
service 3 provided

INFO
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
USR
User-to-user Information
User Information</td></tr>
<tr><td colspan="2">Message flow
<table><tr><td>SIP (Network A)</td><td></td><td>Interconnection Interface</td><td></td><td>SIP (Network B)</td></tr>
<tr><td></td><td></td><td>INVITE (IAM)</td><td>→</td><td></td></tr>
<tr><td></td><td>←</td><td>180 Ringing (ACM)</td><td></td><td></td></tr>
<tr><td></td><td>←</td><td>200 OK INVITE (ANM)</td><td></td><td></td></tr>
<tr><td></td><td></td><td>ACK</td><td>→</td><td></td></tr>
<tr><td></td><td></td><td>INFO (USR)</td><td>→</td><td></td></tr>
<tr><td></td><td>←</td><td>200 OK INFO</td><td></td><td></td></tr>
<tr><td></td><td>←</td><td>INFO (USR)</td><td></td><td></td></tr>
<tr><td></td><td></td><td>200 OK INFO</td><td>→</td><td></td></tr>
<tr><td></td><td></td><td>Apply post test routine</td><td></td><td></td></tr></table></td></tr>
<tr><td>Comments</td><td>Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request and the
a User-to-user Indicator parameter is set to Is the Request service 3 'not essential' or 'essential'?
Check: Is an ISUP/BICC ANM encapsulated in the 200 OK INVITE and the User-to-user Indicator parameter is set to 'Response', 'service 3 provided'?
Check: Is an ISUP/BICC USR encapsulated in the INFO message sent from network A to network B containing an User-to-user Information parameter?
Check: Is an ISUP/BICC USR encapsulated in the INFO message sent from network B to network A containing an User-to-user Information parameter?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_uus_013
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1, 6.5/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 3 not essential rejected in 200 OK response.** BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 3 not essential calls user B not subscribed to User-to-User service 3 the call is rejected by the network an **User-to-user Indicator** parameter is present set to 'Response', 'service 3 not provided' in the encapsulated ANM of the 200 OK final response.
Configuration	User A is subscribed to the User-to-User service 3 User B is not subscribed to the User-to-User service 3
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Indicator Request service 3 not essential 200 OK Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** User-to-user Indicator Response service 3 not provided
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE (IAM) → ← 180 Ringing (ACM) ← 200 OK INVITE (ANM) ACK → **Apply post test routine**
Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request? **Check:** Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 3' 'not essential'? **Check:** Is an ISUP/BICC ANM encapsulated in the 200 OK response? **Check:** Is an User-to-user Indicator parameter present set to 'Response', 'service 3 not provided' in the encapsulated ISUP/BICC ANM? Repeat this test in reverse direction.

Test case number	SS_uus_014
Test case group	SIP-SIP/SIP-I/UUS
Reference	6.11.2, 7.1/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 3 essential rejection.** BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 3 essential calls user B not subscribed to User-to-User service 3 the call is rejected by the network. A 500 Server Internal Error is sent and an encapsulated ISUP/BICC REL is present, the Cause value is set to #29 or #69.
Configuration	User A is subscribed to the User-to-User service 3
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** User-to-user Indicator Request service 3 essential 500 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **REL** Cause value #29 or #69
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE (IAM)	→	
	←	500 Server Internal Error (REL)		
		ACK	→	
		Apply post test routine		

Comments	**Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request? **Check:** Is a User-to-user Indicator parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 1', 'essential'? **Check:** Is an ISUP/BICC REL encapsulated in the 500 response? **Check:** Is the Cause value set to #29 or #69 in the encapsulated REL? Repeat this test in reverse direction.

Test case number	SS_uus_015
Test case group	SIP-SIP/SIP-I/UUS
Reference	5.4.3.2, 6.5, 7.1/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 3 during a session is established successful.** BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 3 user A is able to request the User-to-User service 3 while the session is established. The User-to-User service is successful.
Configuration	User A is subscribed to the User-to-User service 3 User B is subscribed to the User-to-User service 3
SIP Parameter	INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **FAR** Facility indicator user-to-user service User-to-user Indicator Request service 3 not essential INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **FAA** Facility indicator user-to-user service User-to-user Indicator Response service 3 provided INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **USR** User-to-user Information User Information

Message flow

SIP (Network A)		Interconnection Interface		SIP (Network B)
		A session is already established		
		INFO (FAR)	→	
	←	200 OK INFO		
	←	INFO (FAA)		
		200 OK INFO	→	
		INFO (USR)	→	
	←	200 OK INFO		
	←	INFO (USR)		
		200 OK INFO	→	
		Apply post test routine		

Comments	A session is already established **Check:** Is an ISUP/BICC FAR encapsulated in the INFO request sent from Network A to Network B and the a User-to-user Indicator parameter is set to Is the **Request service 3** 'not essential'? **Check:** Is an ISUP/BICC FAA encapsulated in the INFO request sent from Network B to Network A and the User-to-user Indicator parameter is set to 'Response', 'service 3 provided'? **Check:** Is an ISUP/BICC USR encapsulated in the INFO message sent from network A to network B containing an User-to-user Information parameter? **Check:** Is an ISUP/BICC USR encapsulated in the INFO message sent from network B to network A containing an User-to-user Information parameter? Repeat this test in reverse direction.

Test case number	SS_uus_016
Test case group	SIP-SIP/SIP-I/UUS
Reference	5.4.3.2, 6.5, 7.1/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	**SIP-I support: Indicating of User-to-User service 3 during a session is established unsuccessful.** BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 3 user A is able to request the User-to-User service 3 while the session is established. The service request is rejected by Network B.
Configuration	User A is subscribed to the User-to-User service 3 User B is not subscribed to the User-to-User service 3
SIP Parameter	INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **FAR** Facility indicator user-to-user service User-to-user Indicator Request service 3 not essential INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **FRJ** Facility indicator user-to-user service User-to-user Indicator Response service 3 not provided
Message flow	

SIP (Network A)	**Interconnection Interface**	**SIP (Network B)**
	A session is already established	
	INFO (FAR) →	
	← 200 OK INFO	
	← INFO (FRJ)	
	200 OK INFO →	
	Apply post test routine	

Comments	A session is already established **Check:** Is an ISUP/BICC FAR encapsulated in the INFO request sent from Network A to Network B and the a User-to-user Indicator parameter is set to Is the **Request service 3** 'not essential'? **Check:** Is an ISUP/BICC FAA encapsulated in the INFO request sent from Network B to Network A and the User-to-user Indicator parameter is set to 'Response', 'service 3 not provided'? Repeat this test in reverse direction.

7.1.6.2 Subaddressing (SUB)

Test case number	SS_sub_001
Test case group	SIP-SIP/SIP-I/SUB
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 62
Test purpose	**SIP-I support: Calling party subaddress can be correctly transferred in the Access Transport parameters.** BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure that an ISUP/BICC ATP parameter present in the encapsulated IAM of the INVITE request and contains a Calling party subaddress.
Configuration	User A is subscribed to the SUB supplementary service
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Access transport Calling party subaddress --[any boundary name]--
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE(**IAM**) →	
Comments	Establish a call from User A subscribed to the SUB supplementary service to user B **Check:** Is an ISUP/BICC IAM present in the initial INVITE request? **Check:** Is an ISUP/BICC ATP parameter present in the encapsulated IAM containing a Calling party subaddress? Repeat this test in reverse direction.

Test case number	SS_sub_002
Test case group	SIP-SIP/SIP-I/SUB
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 62
Test purpose	**SIP-I support. Called party subaddress can be correctly transferred in the Access Transport parameters.** BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure that an ISUP/BICC ATP parameter present in the encapsulated IAM of the INVITE request and contains a Called party subaddress.
Configuration	User A is subscribed to the SUB supplementary service
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **IAM** Access transport Called party subaddress --[any boundary name]--
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE(**IAM**) → **Apply post test routine**	
Comments	**Check:** Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response? **Check:** Is an ISUP/BICC ATP parameter present in the encapsulated ANM containing a Called party subaddress? Repeat this test in reverse direction.

Test case number	SS_sub_003
Test case group	SIP-SIP/SIP-I/SUB
Reference	6.7/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 62
Test purpose	**SIP-I support. Connected party subaddress can be correctly transferred in the Access Transport parameters.** BICC/ISUP - SIP-I interworking applies in the terminating network User A is located in network A and user B is located in network B. Ensure that an ISUP/BICC ATP parameter present in the encapsulated ANM of the 200 OK INVITE final response and a Connected party subaddress is contained.
Configuration	User B is subscribed to the SUB supplementary service
SIP Parameter	200 OK INVITE Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **ANM** Access transport Connected party subaddress
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE(IAM) → ← 180 Ringing(ACM) ← 200 OK INVITE(**ANM**) ACK → **Apply post test routine**
Comments	**Check:** Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response? **Check:** Is an ISUP/BICC ATP parameter present in the encapsulated ANM containing a Called party subaddress? Repeat this test in reverse direction.

7.1.6.3 Terminal Portability (TP)

Test case number	SS_tp_001
Test case group	SIP-SIP/SIP-I/TP
Reference	5.4.3.2/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 64
Test purpose	**SIP-I support. SUS and RES messages transferred in an INFO request.** BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. A session is already established. Ensure that an INFO request is sent from Network A to Network B and an ISUP SUS message is encapsulated containing a Suspend/resume indicator set to ISDN subscriber initiated. Ensure that an INFO request is sent from Network A to Network B and an ISUP RES message is encapsulated containing a Suspend/resume indicator set to ISDN subscriber initiated.
Configuration	User A is subscribed to the Terminal Portability supplementary service
SIP Parameter	INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **SUS** Suspend/resume indicator ISDN subscriber initiated INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required **RES** Suspend/resume indicator ISDN subscriber initiated
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		A confirmed session already exists		
		INFO(**SUS**)	→	
	←	200 OK INFO		
		INFO(**RES**)	→	
	←	200 OK INFO		
		Apply post test routine		

Comments	A session is already established **Check:** Is an ISUP **SUS** message encapsulated in the INFO request and the Suspend/resume indicator set to 'ISDN subscriber initiated'? **Check:** Is an ISUP **RES** message encapsulated in the INFO request and the Suspend/resume indicator set to 'ISDN subscriber initiated'? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_tp_002</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/SIP-I/TP</td></tr>
<tr><td>Reference</td><td>5.4.3.2, 6.11.2, 6.11.2/[24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 17 AND SE 47 AND SE 64</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. SUS message transferred in an INFO request call released.

BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. A session is already established. Ensure that an INFO request is sent from Network A to Network B and an ISUP SUS message is encapsulated containing a Suspend/resume indicator set to ISDN subscriber initiated. Ensure that an BYE request is sent from Network A to Network B and an ISUP REL message is encapsulated containing a Cause value set to #102.</td></tr>
<tr><td>Configuration</td><td>User A is subscribed to the Terminal Portability supplementary service</td></tr>
<tr><td>SIP Parameter</td><td>INFO
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
SUS
Suspend/resume indicator
ISDN subscriber initiated

BYE
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
REL
Location
public network serving remote user
Cause value
102</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
A confirmed session already exists
INFO(SUS) →
← 200 OK INFO

BYE(REL) →
← 200 OK BYE</td></tr>
<tr><td>Comments</td><td>A session is already established
Check: Is an ISUP SUS message encapsulated in the INFO request and the Suspend/resume indicator set to ISDN 'subscriber initiated'?
Check: Is an ISUP REL message encapsulated in the BYE request and the Cause value set to #102?
Repeat this test in reverse direction.</td></tr>
</table>

7.2 Number Portability

Test case number	SS_NP_001
Test case group	SIP-SIP/NubP
Reference	5.3, 5.4/[2]
SELECTION EXPRESSION	[Network A] SE 13
Test purpose	**Request line in the INVITE contains the number portability indication.** User A attempts to call user B ported to network B. Ensure that the userinfo in the INVITE contains a destination number in the global number format, an 'rn' parameter containing the Number Portability Routing Number in a global number format with hex digits and optional the 'npdi' parameter.
Configuration	
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>[;npdi][; rn=(Number portability routing number)] @<hostname>;user = phone SIP/2.0
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**
Comments	**Check:** Is the URI in the userinfo of the Request line in a global number format? **Check:** Is the URI rn parameter containing the Number Portability Routing Number in a global number format? **Check:** Is optional the URI parameter 'npdi' present? **Check:** Is the user parameter set to 'phone'? Repeat this test in reverse direction.

Test case number	SS_NP_002
Test case group	SIP-SIP/NubP
Reference	5.3, 5.4/[2]
SELECTION EXPRESSION	NOT [Network A] SE 13
Test purpose	**Request line in the INVITE without npdi parameter.** The Network A does not have a Number Portability database. User A attempts to call user B ported to network B. Ensure that the userinfo in the INVITE contains a destination number in a global number format and a npdi URI parameter is not present.
Configuration	
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>@<hostname>;user = phone SIP/2.0
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**
Comments	**Check:** Is the URI in the userinfo of the Request line in a global number format without npdi parameter and number portability routing number? **Check:** Is the user parameter set to 'phone'? Repeat this test in reverse direction.

7.3 Accounting

Test case number	SS_acc_001
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	**Comparison of Charging Data Records > 1 s.** Accounting of a confirmed session with a duration > 1 s. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE	→	
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
		BYE	→	
	←	200 OK BYE		

Comments	1. Setup a call from network A to network B. 2. Verify is the session confirmed. 3. Terminate the session after 5 s. 4. Determine the duration of the session from the trace of the call monitor. 5. Compare the following information elements indicated in the CDR´s of both networks: • calling party number • called party number • timestamp • callduration • callsetuptime (optional) 6. Check the duration indicated in the CDR against the duration in the call trace. 7. Repeat this test in reverse direction.

Test case number	SS_acc_002
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	**Comparison of Charging Data Records < 1 s** Accounting of a confirmed session with a duration of < 1 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.
Configuration	
SIP Parameter	
Message flow	
SIP (Network A)	**Interconnection Interface** / **SIP (Network B)**
	INVITE →
	← 180 Ringing
	← 200 OK INVITE
	ACK →
	Communication
	BYE →
	← 200 OK BYE
Comments	1. Setup a call from network A to network B. 2. Verify is the session confirmed. 3. Terminate the session after 5 s. 4. Determine the duration of the session from the trace of the call monitor. 5. Compare the following information elements indicated in the CDR´s of both networks: • calling party number • called party number • timestamp • callduration • callsetuptime (optional) 6. Check the duration indicated in the CDR against the duration in the call trace. 7. Repeat this test in reverse direction.

Test case number	SS_acc_003
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	**Comparison of Charging Data Records > 15 min.** Accounting of a confirmed session with a duration of > 15 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE	→	
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		Communication		
		BYE	→	
	←	200 OK BYE		

Comments	1. Setup a call from network A to network B. 2. Verify is the session confirmed. 3. Terminate the session after 15 min. 4. Determine the duration of the session from the trace of the call monitor. 5. Compare the following information elements indicated in the CDR´s of both networks: • calling party number • called party number • timestamp • callduration • callsetuptime (optional) 6. Check the duration indicated in the CDR against the duration in the call trace. 7. Repeat this test in reverse direction.

Test case number	SS_acc_004
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	**Comparison of Charging Data Records 25 min.** Accounting of a confirmed session with a duration of 25 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing ← 200 OK INVITE ACK → **Communication** BYE → ← 200 OK BYE	
Comments	1. Setup a call from network A to network B. 2. Verify is the session confirmed. 3. Terminate the session after 25 min. 4. Determine the duration of the session from the trace of the call monitor. 5. Compare the following information elements indicated in the CDR´s of both networks: • calling party number • called party number • timestamp • callduration • callsetuptime (optional) 6. Check the duration indicated in the CDR against the duration in the call trace. 7. Repeat this test in reverse direction.

Test case number	SS_acc_005
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	**Comparison of Charging Data Records more than 30 min.** Accounting of a confirmed session with a duration of > 30 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing ← 200 OK INVITE ACK → **Communication** BYE → ← 200 OK BYE
Comments	1. Setup a call from network A to network B. 2. Verify is the session confirmed. 3. Terminate the session after 35 min. 4. Determine the duration of the session from the trace of the call monitor. 5. Compare the following information elements indicated in the CDR´s of both networks: • calling party number • called party number • timestamp • callduration • callsetuptime (optional) 6. Check the duration indicated in the CDR against the duration in the call trace. 7. Repeat this test in reverse direction.

Test case number	SS_acc_006
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	**Comparison of Charging Data Records more than 60 min.** Accounting of a confirmed session with a duration between 60 min and 120 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.
Configuration	
SIP Parameter	
Message flow	**SIP (Network A)** — **Interconnection Interface** — **SIP (Network B)** INVITE → ← 180 Ringing ← 200 OK INVITE ACK → **Communication** BYE → ← 200 OK BYE
Comments	1. Setup a call from network A to network B. 2. Verify is the session confirmed. 3. Terminate the session at the earliest 61 min and at the latest 119 min. 4. Determine the duration of the session from the trace of the call monitor. 5. Compare the following information elements indicated in the CDR´s of both networks: • calling party number • called party number • timestamp • callduration • callsetuptime (optional) 6. Check the duration indicated in the CDR against the duration in the call trace. 7. Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_acc_007</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/ACCOUNTING</td></tr>
<tr><td>Reference</td><td></td></tr>
<tr><td>SELECTION EXPRESSION</td><td></td></tr>
<tr><td>Test purpose</td><td>Comparison of Charging Data Records more than 24 hours.

Accounting of a confirmed session with duration > 24 h with change of date. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td></td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
INVITE →
← 180 Ringing
← 200 OK INVITE
ACK →
Communication
BYE →
← 200 OK BYE</td></tr>
<tr><td>Comments</td><td>1. Setup a call from network A to network B.
2. Verify is the session confirmed.
3. Terminate the session after 24 hours.
4. Determine the duration of the session from the trace of the call monitor.
5. Compare the following information elements indicated in the CDR´s of both networks:
• calling party number
• called party number
• timestamp
• callduration
• callsetuptime (optional)
6. Check the duration indicated in the CDR against the duration in the call trace.
7. Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_acc_008
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	**Comparison of Charging Data Records less than 1 s.** Accounting of a confirmed session with duration <1 s. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing ← 200 OK INVITE ACK → **Communication** BYE → ← 200 OK BYE
Comments	1. Setup a call from network A to network B. 2. Verify is the session confirmed. 3. Terminate the session after 0,9 s. 4. Determine the duration of the session from the trace of the call monitor. 5. Compare the following information elements indicated in the CDR´s of both networks: • calling party number • called party number • timestamp • callduration • callsetuptime (optional) 6. Check the duration indicated in the CDR against the duration in the call trace. 7. Repeat this test in reverse direction.

Test case number	SS_acc_009
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	**Comparison of Charging Data Records session not confirmed.** Accounting of an unsuccessful session in the early dialogue. Verify the duration of the call attempt stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface if applicable.
Configuration	
SIP Parameter	
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)** INVITE → ← 180 Ringing BYE/CANCEL → ← 200 OK BYE/CANCEL ← 487 Request Terminated ACK →	
Comments	1. Setup a call from network A to network B. 2. Verify is an early dialogue established. 3. Terminate the early dialogue after 20 s. 4. Determine the duration of the session from the trace of the call monitor. 5. Compare the following information elements indicated in the CDR´s of both networks: • calling party number • called party number • timestamp • callduration • callsetuptime (optional) 6. Check the duration indicated in the CDR against the duration in the call trace. 7. Repeat this test in reverse direction.

7.4 Carrier Selection

Test case number	SS_csel_001
Test case group	SIP-SIP/CS
Reference	5.7.1.10/[2]
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15
Test purpose	**User selects an operator 'call-by-call'.** User A and user B are located in network A. Ensure that user A is able to call user B and user A is able to select network B as a selected carrier 'call-by-call'.
Configuration	User in network A is not presubscribed
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0 INVITE: Request line sip: + <CC> <NDC> <SN>;npdi [;rn=<Number portability routing number>]@<hostname>; user=phone SIP/2.0
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE 1 → ← INVITE 2 **Apply post test routine**
Comments	**Check:** Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? **Check:** Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? **Check:** Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? **NOTE 1:** The 'cic' parameter may be absent according national regulations or national agreements. **NOTE 2:** It is possible that further informations are available in the Request line regarding the end user charging in case of Carrier selection. Repeat this test in reverse direction.

Test case number	SS_csel_002
Test case group	SIP-SIP/CS
Reference	5.7.1.10/[2]
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15
Test purpose	**User is presubscribed to operator B.** User A and user B are located in network A. Ensure that user A is able to call user B and user A is preselected to network B as a selected carrier.
Configuration	User in network A is presubscribed to network B
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0 INVITE: Request line sip: + <CC> <NDC> <SN>;npdi [;rn=<Number portability routing number>]@<hostname>; user=phone SIP/2.0
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE 1 → ← INVITE 2 **Apply post test routine**
Comments	**Check:** Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? **Check:** Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? **Check:** Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? **NOTE 1:** The 'cic' parameter may be absent according national regulations or national agreements. **NOTE 2:** It is possible that further informations are available in the Request line regarding the end user charging in case of Carrier selection. Repeat this test in reverse direction.

Test case number	SS_csel_003
Test case group	SIP-SIP/CS
Reference	5.7.1.10/[2]
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15
Test purpose	**User is presubscribed to an operator unequal to B, and overrides the preselection with call-by-call via operator B.** User A and user B are located in network A. User A is preselected to a network unequal to network B. Ensure that user A is able to call user B and user A is able to select network B as a selected carrier 'call-by-call'. The preselected carrier is ignored.
Configuration	User in network A is presubscribed to network B
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0 INVITE: Request line sip: + <CC> <NDC> <SN>;npdi [;rn=<Number portability routing number>]@<hostname>; user=phone SIP/2.0

Message flow

SIP (Network A)		**Interconnection Interface**		**SIP (Network B)**
		INVITE 1	→	
	←	INVITE 2		
		Apply post test routine		

Comments	**Check:** Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? **Check:** Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? **Check:** Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? **NOTE 1:** The 'cic' parameter may be absent according national regulations or national agreements. **NOTE 2:** It is possible that further informations are available in the Request line regarding the end user charging in case of Carrier selection. Repeat this test in reverse direction.

Test case number	SS_csel_004
Test case group	SIP-SIP/CS
Reference	5.7.1.10/[2]
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15
Test purpose	**User is presubscribed to an operator not operator B, and overrides the preselection with call-by-call via operator B.** User A and user B are located in network A. User A is preselected to a network unequal to network B. Ensure that user A is able to call user B and user A is able to select network B as a selected carrier 'call-by-call'. The preselected carrier is ignored.
Configuration	User in network A is presubscribed **not** to network B
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0 INVITE: Request line sip: + <CC> <NDC> <SN>;npdi [;rn=<Number portability routing number>]@<hostname>; user=phone SIP/2.0
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE 1 → ← INVITE 2 **Apply post test routine**
Comments	**Check:** Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? **Check:** Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? **Check:** Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? **NOTE 1:** The 'cic' parameter may be absent according national regulations or national agreements. **NOTE 2:** It is possible that further informations are available in the Request line regarding the end user charging in case of Carrier selection. Repeat this test in reverse direction.

Test case number	SS_csel_005
Test case group	SIP-SIP/CS
Reference	
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15 AND [Network A] SE34
Test purpose	**User is preselected to operator B. Transit of CUG information -OA.** An originating user in a CUG Outgoing Access not allowed preselected to Network B and calls to a user in the same CUG. The session establishment is successful.
Configuration	User in network A is presubscribed to network B Users in network A are in the same CUG
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>@tariff.<hostname> user=phone SIP/2.0 Content-Type: application/vnd.etsi.cug+xml Content-Disposition: ….;handling= required ……. <…:cug> …… <..: cugCommunicationIndicator>11</…: cugCommunicationIndicator> <…:cug> INVITE: Request line sip: + <CC> <NDC> <SN@<hostname>;user=phone SIP/2.0 Content-Type: application/vnd.etsi.cug+xml Content-Disposition: ….;handling= required ……. <…:cug> …… <..: cugCommunicationIndicator>11</…: cugCommunicationIndicator> <…:cug>
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE 1 → ← INVITE 2 **Apply post test routine**
Comments	**Check:** Is the sub domain pattern 'tariff' present at the beginning of the hostportion only of the initial INVITE sent from network A to network B? **Check:** Is the 'npdi' parameter present in the userinfo of the INVITE request sent from network B to network A? **Check:** Is optional the 'rn' parameter present in the userinfo of the INVITE request sent from network B to network A? **Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? **Check:** Is the session setup not rejected?

7.5 Emergency call

Test case number	SS_ecall_001
Test case group	SIP-SIP/EmC
Reference	5.2.10, 5.7.1.14/[2]
SELECTION EXPRESSION	
Test purpose	**Request line in the INVITE.** User A attempts to call a PSAP located in network B. Ensure that the Request line in the INVITE contains the emergency number and a 'rn' parameter containing the PSAP routing number. In addition a location information may be present: • Geolocation header • P-Access-Network-Info header • National solution to convey location information to make location information available for the PASP.
Configuration	
SIP Parameter	INVITE: Request line sip+ <(emergency number)>[; rn =+<(PASP routing number)] @hostname>;user = phone SIP/2.0
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE ➔ **Apply post test routine**
Comments	**Check:** Is the URI in the userinfo of the Request line in a global number format containing the PSAP routing number? **Check:** Optional: Is the URI 'rn' parameter containing the PASP Routing Number? **Check:** Is the user parameter set to 'phone'? Repeat this test in reverse direction.

7.6 SIP Support of Charging

<table>
<tr><td>Test case number</td><td colspan="3">SS_sipc_001</td></tr>
<tr><td>Test case group</td><td colspan="3">SIP-SIP/ SIP_charging</td></tr>
<tr><td>Reference</td><td colspan="3">B.2.3/[19]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td colspan="3">SE 16</td></tr>
<tr><td>Test purpose</td><td colspan="3">Successful session from user A to user B via network B one single tariff.

User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a tariff information with one single tariff covered in a XML MIME body in a reliable provisional or successful final response.</td></tr>
<tr><td>Configuration</td><td colspan="3"></td></tr>
<tr><td>SIP Parameter</td><td colspan="3">INVITE:
Supported: 100rel

18x or 200 OK
Require: 100rel
ContentType: application/vnd.etsi.sci+xml
Content-Disposition: render; handling=optional

messageType
crgt
chargingControlIndicators
chargingTariff
tariffCurrency
currentTariffCurrency
communicationChargeSequenceCurrency
currencyFactorScale
currencyFactor
currencyScale
tariffDuration
subTariffControl
tariffControlIndicators
originationIdentification
currency (optional)</td></tr>
<tr><td>Message flow
SIP (Network A)</td><td></td><td>Interconnection Interface</td><td>SIP (Network B)</td></tr>
<tr><td></td><td></td><td>INVITE</td><td>→</td></tr>
<tr><td>CASE A</td><td>←</td><td>18x(crgt)</td><td></td></tr>
<tr><td></td><td></td><td>PRACK</td><td>→</td></tr>
<tr><td></td><td>←</td><td>200 OK PRACK</td><td></td></tr>
<tr><td>CASE B</td><td>←</td><td>200 OK INVITE(crgt)</td><td></td></tr>
<tr><td></td><td></td><td>Apply post test routine</td><td></td></tr>
<tr><td>Comments</td><td colspan="3">Check: Is the supported header in the initial INVITE set to '100rel'
Check: Is the Require header in the response containing the tariff information set to '100rel'?
Check: Is the messageType 'crgt' present in a 1xx provisional or a 200 OK INVITE final response?
Check: Is the tariffCurrency element set to 'currentTariffCurrency'?
Check: Represents the currencyFactorScale in the communicationChargeSequenceCurrency element the applicable tariff?
Check: Is the tariffDuration element set to '0'?
Check: Is the optional element 'currency' set to 'EUR' if present?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_sipc_002
Test case group	SIP-SIP/ SIP_charging
Reference	B.2.3/[19]
SELECTION EXPRESSION	SE 16
Test purpose	**Successful session from user A to user B via network B several tariffs in one sequence.** User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a tariff information with several tariffs in a sequence covered in a XML MIME body in a reliable provisional or successful final response.
Configuration	

SIP Parameter	INVITE:
	    Supported: 100rel
	
	18x or 200 OK
	    Require: 100rel
	    ContentType: application/vnd.etsi.sci+xml
	    Content-Disposition: render; handling=optional
	
	    messageType
	    crgt
	        chargingControlIndicators
	        chargingTariff
	            tariffCurrency
	                currentTariffCurrency
	                    communicationChargeSequenceCurrency
	                        currencyFactorScale
	                            currencyFactor
	                            currencyScale
	                        tariffDuration
	                        subTariffControl
	                    communicationChargeSequenceCurrency
	                        currencyFactorScale
	                            currencyFactor
	                            currencyScale
	                        tariffDuration
	                        subTariffControl
	                    tariffControlIndicators
	        originationIdentification
	        currency (optional)

Message flow	
SIP (Network A)	**Interconnection Interface** → **SIP (Network B)**
	INVITE →
CASE A	← 18x(crgt)
	PRACK →
	← 200 OK PRACK
CASE B	← 200 OK INVITE(crgt)
	Apply post test routine
Comments	**Check:** Is the Supported header in the initial INVITE set to '100rel'? **Check:** Is the Require header in the response containing the tariff information set to '100rel'? **Check:** Is the messageType '**crgt**' present in a 1xx provisional or a 200 OK INVITE final response? **Check:** Is the **tariffCurrency** element set to 'currentTariffCurrency'? **Check:** Are there **more than one communicationCharge SequenceCurrency** elements present in the currentTariffCurrency element? **Check:** Represents the **currencyFactorScale** in the communicationCharge SequenceCurrency elements the applicable tariffs? **Check:** Is the tariffDuration element in the last applicable tariff set to '0'? **Check:** Is the optional element '**currency**' set to 'EUR' if present? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_sipc_003</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/ SIP_charging</td></tr>
<tr><td>Reference</td><td>B.2.3/[19]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>SE 16</td></tr>
<tr><td>Test purpose</td><td>Successful session from user A to user B via network B with call attempt charge.

User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a tariff information with a call attempt charge covered in a XML MIME body in a reliable provisional or successful final response.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td>INVITE:
Supported: 100rel

18x or 200 OK
Require: 100rel
ContentType: application/vnd.etsi.sci+xml
Content-Disposition: render; handling=optional

messageType
crgt
chargingControlIndicators
chargingTariff
tariffCurrency
currentTariffCurrency
communicationChargeSequenceCurrency
currencyFactorScale
currencyFactor
currencyScale
tariffDuration
subTariffControl
tariffControlIndicators
callAttemptChargeCurrency
currencyFactor
currencyScale
originationIdentification
currency (optional)</td></tr>
<tr><td colspan="2">Message flow
<table>
<tr><td>SIP (Network A)</td><td></td><td>Interconnection Interface</td><td></td><td>SIP (Network B)</td></tr>
<tr><td></td><td></td><td>INVITE</td><td>→</td><td></td></tr>
<tr><td>CASE A</td><td>←</td><td>18x(crgt)</td><td></td><td></td></tr>
<tr><td></td><td></td><td>PRACK</td><td>→</td><td></td></tr>
<tr><td></td><td>←</td><td>200 OK PRACK</td><td></td><td></td></tr>
<tr><td>CASE B</td><td>←</td><td>200 OK INVITE(crgt)</td><td></td><td></td></tr>
<tr><td></td><td></td><td>Apply post test routine</td><td></td><td></td></tr>
</table></td></tr>
<tr><td>Comments</td><td>Check: Is the supported header in the initial INVITE set to '100rel'?
Check: Is the Require header in the response containing the tariff information set to '100rel'?
Check: Is the messageType a 'crgt' present in a 1xx provisional or a 200 OK INVITE final response?
Check: Is the tariffCurrency element set to 'callAttemptChargeCurrency'?
Check: Represents the currencyFactorScale in the callAttemptChargeCurrency element the applicable tariff?
Check: Is the optional element 'currency' set to 'EUR' if present?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_sipc_004
Test case group	SIP-SIP/ SIP_charging
Reference	B.2.3/[19]
SELECTION EXPRESSION	SE 16
Test purpose	**Successful session from user A to user B via network B with call setup charge.** User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a tariff information with a call setup charge covered in a XML MIME body in a reliable provisional or successful final response.
Configuration	
SIP Parameter	INVITE: Supported: 100rel 18x or 200 OK Require: 100rel ContentType: application/vnd.etsi.sci+xml Content-Disposition: render; handling=optional messageType crgt chargingControlIndicators chargingTariff tariffCurrency currentTariffCurrency communicationChargeSequenceCurrency currencyFactorScale currencyFactor currencyScale tariffDuration subTariffControl tariffControlIndicators callSetupChargeCurrency currencyFactor currencyScale originationIdentification currency (optional)
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** INVITE → **CASE A** ← 18x(crgt) PRACK → ← 200 OK PRACK **CASE B** ← 200 OK INVITE(crgt) **Apply post test routine**
Comments	**Check:** Is the supported header in the initial INVITE set to '100rel'? **Check:** Is the Require header in the response containing the tariff information set to '100rel'? **Check:** Is the messageType a '**crgt**' present in a 1xx provisional or a 200 OK INVITE final response? **Check:** Is the **tariffCurrency** element set to 'callSetupChargeCurrency'? **Check:** Represents the **currencyFactorScale** in the **callSetupChargeCurrency** element the applicable tariff? **Check:** Is the optional element '**currency**' set to 'EUR' if present? Repeat this test in reverse direction.

Test case number	SS_sipc_005
Test case group	SIP-SIP/ SIP_charging
Reference	B.2.3/[19]
SELECTION EXPRESSION	SE 16
Test purpose	**Successful session from user A to user B via network B with a next tariff.** User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a tariff information with a next tariff and tariff switch over time covered in a XML MIME body in a reliable provisional or successful final response.
Configuration	
SIP Parameter	INVITE: Supported: 100rel 18x or 200 OK Require: 100rel ContentType: application/vnd.etsi.sci+xml Content-Disposition: render; handling=optional messageType crgt chargingControlIndicators chargingTariff tariffCurrency currentTariffCurrency communicationChargeSequenceCurrency currencyFactorScale currencyFactor currencyScale tariffDuration subTariffControl tariffControlIndicators tariffSwitchCurrency nextTariffCurrency communicationChargeSequenceCurrency currencyFactorScale currencyFactor currencyScale tariffDuration subTariffControl tariffControlIndicators tariffSwitchOverTime originationIdentification currency (optional)
Message flow **SIP (Network A)** **Interconnection Interface** **SIP (Network B)**	INVITE → **CASE A** ← 18x(crgt) PRACK → ← 200 OK PRACK **CASE B** ← 200 OK INVITE(crgt) **Apply post test routine**
Comments	**Check:** Is the supported header in the initial INVITE set to '100rel'? **Check:** Is the Require header in the response containing the tariff information set to '100rel'? **Check:** Is the messageType '**crgt**' present in a 1xx provisional or a 200 OK INVITE final response? **Check:** Is the **tariffSwitchCurrency** element set to 'nextTariffCurrency'? **Check:** Represents the **currencyFactorScale** in the communicationChargeSequenceCurrency element the next tariff? **Check:** Is the time to change the tariff indicated in the **tariffSwitchOverTime** element? **Check:** Is the optional element '**currency**' set to 'EUR' if present? Repeat this test in reverse direction.

Test case number	SS_sipc_006
Test case group	SIP-SIP/ SIP_charging
Reference	B.2.3/[19]
SELECTION EXPRESSION	SE 16
Test purpose	**Successful change of a current tariff and next tariff during an active session.** User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a new tariff information with several current tariffs and several next tariffs covered in a XML MIME body in an INFO request.
Configuration	
SIP Parameter	INFO ContentType: application/vnd.etsi.sci+xml messageType crgt chargingControlIndicators chargingTariff tariffCurrency currentTariffCurrency communicationChargeSequenceCurrency currencyFactorScale currencyFactor currencyScale tariffDuration subTariffControl communicationChargeSequenceCurrency currencyFactorScale currencyFactor currencyScale tariffDuration subTariffControl tariffControlIndicators tariffSwitchCurrency nextTariffCurrency communicationChargeSequenceCurrency currencyFactorScale currencyFactor currencyScale tariffDuration subTariffControl communicationChargeSequenceCurrency currencyFactorScale currencyFactor currencyScale tariffDuration subTariffControl tariffControlIndicators tariffSwitchOverTime originationIdentification currency (optional)
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** **A confirmed session already exists** ← INFO 200 OK INFO → **Apply post test routine**
Comments	**Check:** Is the messageType '**crgt**' present in the INFO request? **Check:** Is the **tariffCurrency** element set to 'currentTariffCurrency'? **Check:** Represents the **currencyFactorScale** in the communicationChargeSequenceCurrency elements the current tariffs? **Check:** Is the **tariffSwitchCurrency** element set to 'nextTariffCurrency'? **Check:** Represents the **currencyFactorScale** in the communicationChargeSequenceCurrency elements the next tariffs? Repeat this test in reverse direction.

Test case number	SS_sipc_007
Test case group	SIP-SIP/SIP_charging
Reference	B.2.3/[19]
SELECTION EXPRESSION	SE 16
Test purpose	**Successful additional charge during an active session.** User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a new tariff information with additional charge covered in a XML MIME body in an INFO request.
Configuration	
SIP Parameter	INFO ContentType: application/vnd.etsi.sci+xml messageType aocrg chargingControlIndicators addOnCharge addOnChargeCurrency currencyFactor currencyScale originationIdentification currency (optional)
Message flow **SIP (Network A)**	**Interconnection Interface** **SIP (Network B)** **A confirmed session already exists** ← INFO 200 OK INFO → **Apply post test routine**
Comments	**Check:** Is the messageType '**aocrg**' present in the INFO request? **Check:** Is the addOnCharge element set to 'addOnChargeCurrency'? **Check:** Represents the **currencyFactorScale** the add on tariff? Repeat this test in reverse direction

7.7 Quality of Service

7.7.1 Reference Configurations

7.7.1.1 Backbone Configuration

Figure 7.7-1 shows the backbone configuration.

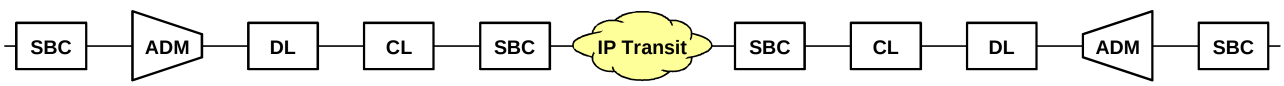

Figure 7.7-1: Backbone

7.7.1.2 PSTN/ISDN classic access Configuration

Figure 7.7-2 shows the PSTN/ISDN classic access configuration.

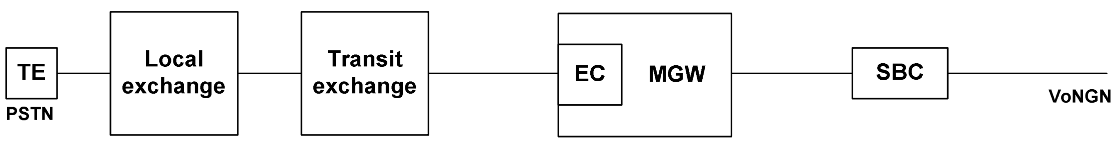

Figure 7.7-2: Reference configuration for PSTN/ISDN with classical access

7.7.1.3 NGN PSTN/ISDN access Configuration

Figure 7.7-3 shows the NGN PSTN/ISDN classic access configuration.

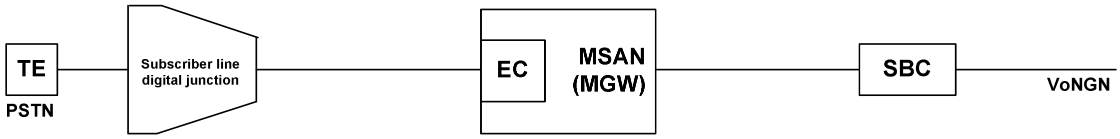

Figure 7.7-3: Reference configuration for NGN with PSTN/ISDN access

7.7.1.4 Access DSL Configuration

Figure 7.7-4 shows the xDSL access configuration.

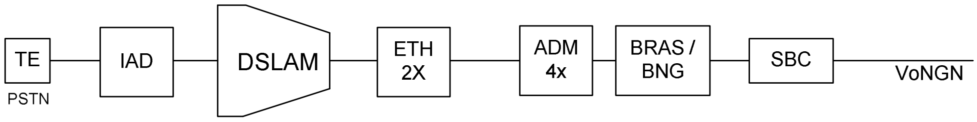

Figure 7.7-4: Reference configuration for DSL access

7.7.1.5 Delay Values

The requirements for the backbone delay, Network parameters: End-to-End Delay, Talker Echo Loudness Rating, R Value Delay with regional propagation delay (1 400 km/11 ms) are contained in clause 4 of TR 102 775 [i.3]

7.7.2 Test purposes for Quality of Service test

Test case number	SS_qos_001
Test case group	SIP-SIP/QoS
Transmission Type:	Voice
Preconditions user segment A:	Reset Jitter Buffer 1 and Jitter Buffer 2 (e.g. by establishing a new call) Apply signal "single-talk" to Interface A and determine Delay D_{JB1} Apply signal "single-talk" to Interface B and determine Delay D_{JB2}
Preconditions user segment B:	Reset Jitter Buffer 1 and Jitter Buffer 2 (e.g. by establishing a new call) Apply signal single-talk to Interface A and determine Delay D_{JB1} Apply signal single-talk to Interface B and determine Delay D_{JB2}
Requirement	$D_{JB1} = D_{JB2}$ Delay jitter for Voice
Test objective	Delay Voice test with loopback
Measurement procedure	After establishing a voice call from the user segment A to user segment B, determine the round trip delay in the sending and receiving direction. Based on the measured delays in the user segment A and user segment B determine the transit segment delay. Loop in user segment B $D_{tr\ seg\ A\text{-}B} = (D_{sum\ seg\ A\text{-}B} - D_{JB1seg\ B} - D_{JB2segA})/2$ Loop in user segment A $D_{tr\ seg\ B\text{-}A} = (D_{sum\ seg\ B\text{-}A} - D_{JB1seg\ B} - D_{JB2segA})/2$
Calling station	The amplitude of the tone is -16 dBm0
Called station	The amplitude of the tone is -16 dBm0
Delay loop	1 000 ms

Test case number	SS_qos_002
Test case group	SIP-SIP/QoS
Transmission Type:	Voice
Preconditions user segment A:	Reset Jitter Buffer 1 and Jitter Buffer 2 (e.g. by establishing a new call) Apply signal "single-talk" to Interface A and determine Delay D_{JB1} and D_{JB2}
Preconditions user segment B:	Reset Jitter Buffer 1 and Jitter Buffer 2 (e.g. by establishing a new call) Apply signal "single-talk" to Interface A and determine Delay D_{JB1} and D_{JB2}
Requirement	$D_{JB1} = D_{JB2}$ Delay jitter for Voice
Test objective	Delay Voice test with synchronous tests system
Measurement procedure	After establishing a voice call from the user segment A to user segment B, determine the delay of the end-to-end in the sending and receiving direction. Based on the measured delays in the user segment A and user segment B determine the transit segment delay. $D_{tr\text{-}seg\ A\text{-}B} = D_{sum\text{-}seg\ A\text{-}B} - D_{JB1seg\ B}$ $D_{tr\text{-}seg\ B\text{-}A} = D_{sum\text{-}seg\ B\text{-}A} - D_{JB2segA}$
Calling station	The amplitude of the tone is -16 dBm0
Called station	The amplitude of the tone is -16 dBm0

Annex A (informative): Bibliography

- IETF RFC 3966 (2004): "The tel URI for Telephone Numbers".
- IETF RFC 3311 (2002): "The Session Initiation Protocol (SIP) UPDATE Method".
- IETF RFC 3323 (2002): "A Privacy Mechanism for the Session Initiation Protocol (SIP)".
- IETF RFC 3325 (2002): "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks".
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History

Document history		
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