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

Reference

DTS/INT-00058

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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.

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	[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

subsystem; Protocol specification (3GPP TS 24.608 Release 10)".

Telecommunications System (UMTS); LTE; Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM) Core Network (CN)

- [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.

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_xx	X
	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	
	CO	IOO_COGI_XXX	
	EmC	SS_ecall_xxx	
	SIP_charging	SS_sipc_xxx	
	SIP-SIP/QoS	SS_qos_xxx	
	311 311 / Q 3 3	100_900_////	

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.

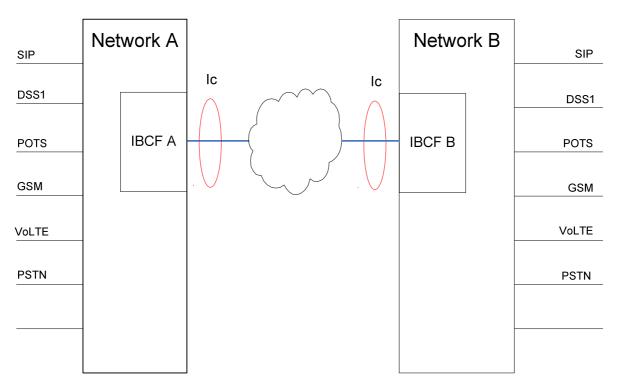


Figure 5.2-1: Reference configuration for the interconnection test

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.

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

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

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	Support
		Network A	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		
3E Z.	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		I.
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		
J_ 2U.	(CFU)?		
SE 26:	The network supports Communication Forwarding Busy (CFB)?		
SE 27:	The network supports Communication Forwarding Busy (CFBR)?		
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		
05.04	(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-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		1
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		
JE 31.	PSTN/PLMN Supplementary services		1
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		
	<mark>←</mark> 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 INVIT		the initial INVITE request.	
	Ensure that the support of the P-Early.Media he INVITE request. A P-Early-Media header is pres		
Configuration		••	
SIP Parameter	INVITE		
	P-Early-Media: supported SDP		
Message flow			
SIP (Network A)	Interconnection Interface INVITE →	SIP (Network B)	
	Apply post test routine		
Comments	Establish a communication from network A to N Check: Is a P-Early-Media header is present		
	Repeat this test in reverse direction.		

Test case number	SS_bcall_007
Test case group	BCALL/successful
Reference	8/[21]
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE3 AND SE 42
Test purpose	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.
Configuration	
SIP Parameter	INVITE P-Early-Media: supported SDP
	183 P-Early-Media: [any value authorizes early media] SDP
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 183 Session Progress Apply post test routine
Comments	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.

Test case number	SS_bcall_008
Test case group	BCALL/successful
Reference	8/[21]
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE 3
Test purpose	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.
Configuration	
SIP Parameter	INVITE P-Early-Media: supported SDP
	180 P-Early-Media: [any value authorizes early media] SDP
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: 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.

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	
Comments	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 a="" ibcf="" in="" network="" of=""></address>
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 a="" ibcf="" in="" network="" of="">; branch=[any value]</address>
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
<u> </u>	- 5 5 5
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 b="" ibcf="" in="" network="" of="">;lr,</address>
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
(A confirmed session already exists
	BYF →
	€ 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 a="" ibcf="" in="" network="" of="">;Ir,</address>	
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 b="" ibcf="" in="" network="" of="">;Ir,</address>
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 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.</dynamic-pt>	
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)</port></port>	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)	
SIF (Network A)	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)	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.
	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	E_SDP	m= line		b= line	a= line
VA	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwidth-value></bandwidth-value></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding</clock></encoding></dynamic-pt>
				(see note)	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</dynamic-pt>
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</dynamic-pt>
VA_05	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> CLEARMODE</dynamic-pt>
NOTE: <bandwidth value=""> for <modifier> of AS is evaluated to be B kbit/s.</modifier></bandwidth>					

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</port></port>
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></dynamic-pt></port>				
	a=rtpmap <dynamic-pt> PCMA/8000</dynamic-pt>				
	a=gpmd; vbd=yes				
	190/200 OK INVITE: SDD				
	180/200 OK INVITE: SDP				
	m=audio <port> RTP/AVP <dynamic-pt> a=rtpmap <dynamic-pt> PCMA/8000</dynamic-pt></dynamic-pt></port>				
	a=tpmap <uynamic-r1> PCMA/80000 a=gpmd; vbd=yes</uynamic-r1>				
Message flow	u-gpina, vou-you				
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	otato and property of a saturation.		
SIP Parameter	INVITE: SDP		
	m=image <port> udptl t38</port>		
	180/200 OK INVITE: SDP		
	m=image <port> udptl t38</port>		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE (SDP1) →		
	← 180 Ringing ← 200 OK INVITE (SDP2)		
	← 200 OK INVITE (SDP2) ACK		
	Apply post test routine		
Comments	Establish a communication from network A to Network B		
Comments	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 met	hod is used.
		that call establishment using <mark>overlap</mark>	
			transfer on the media and B-channels
	is perfor	med correctly.	
Configuration			
SIP Parameter			
Message flow			015 (1) (1 5)
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)	7
		Apply post test routine	
Comments			using the overlap operation in ISDN
	Check:		same Call ID and From header
		values.	
		wers 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 overlage Ensure that in the confirmed state the voice is performed correctly.			
Configuration				
SIP Parameter	INVITE 2: Supported: 100rel			
	183: Require: 100rel			
	INFO: Content-Type: application/x-session-inf SubsequentDigit: <additional digits=""></additional>	io		
Message flow				
SIP (Network A)	Interconnection Interface 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	SIP (Network B) → → →		
	 ← 200 OK INFO ← 180 Ringing(CSq 2) Apply post test routine 			
Comments	Establish a communication from ISDN to S Check: All INVITE requests contains the Check: The 183 session Progress that established Require header set to 100rel. Check: All INFO requests contain the Cosession-info'.	same Call ID and From header values. stablishes an early dialogue contains a intent-Type header set to 'application/x-bubsequentDigit:' MIME body containing		

Test case number	SS_bcall_025		
Test case group	BCALL/st	uccessful	
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 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.		
Configuration		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		
	<pre><?xml version="1.0" encoding="utf-8"?> PSTN BearerCapability BCoctet3</pre>		
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 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.		
	Inchear	113 153 111 1575135 UII56UOII.	

Table 7.1.1-2: PSTN XML BearerCapability

ITC_value	BC Information transfer capability	XML InformationTransferCabability
ITC_VA_1	Speech	<mark>'00000</mark> '
ITC_VA_2	3,1 kHz audio	<mark>'10000'</mark>
ITC VA 3	unrestricted digital information	<mark>'01000</mark> '

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				
	Overal variety IIA OII are analysis III-44 OIIO				
	<pre><?xml version="1.0" encoding="utf-8"?></pre>				
	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.				
	הפףפמו ווווס נפסו ווו ופעפוספ מוופטנוטוו.				

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 CID (Network D)				
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	1 Togicoodicompilotizocco 1110				
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	5.1.1.2/[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				
	20 ml version "1.0" enceding "uff 9"2				
	xml version="1.0" encoding="utf-8"?				
	PSTN Repres Canability				
	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	Territoria de la companya della companya della companya de la companya della comp				
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.				
	1 1				

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			
OID D	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 encoding= uti-8 ?>			
	BearerCapability			
	BCoctet3 CodingStandard>00<			
	InformationTransferCabability>10001<			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE ->			
	← 180 Ringing			
	← 200 OK INVITE			
	ACK →			
Commonto	Apply post test routine Check: Is a PSTN XML MIME body contained in the 200 OK INVITE			
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/s	uccessful			
Reference	5.1.2.3/[2	25]			
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6				
Test purpose	Fall back occurs.				
1 3 5 5 Fair P 3 5 5		. ••••			
	User B is	located in network B and an ISDN end device is used. The Fallback			
		on 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 XI	ML MIME ProgressIndicator body is present, the ProgressDescription is			
	set to '00				
Configuration	User B is an ISDN access either in the PSTN or the SIP - ISDN interworking				
_	according	g [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		1 TogicssDescription>0000 ToTC			
SIP (Network A)		Interconnection Interface SIP (Network B)			
on (notwork /t)		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			
	L	codec list in the SDP?			
	Check:	Is the InformationTransferCabability element value consistent with the			
		bandwidth information in the SDP?			
	Repeat th	nis 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.			
rest purpose	on Toupport, Busic can, IAM present in the INTTL 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			
	TARA			
	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)			
	nocess transport (optional)			
	[any boundary name]			
Message flow	[any soundary name]			
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.					
0	Ensure that the original IAM is encapsulated in any INVITE request.					
Configuration						
SIP Parameter						
Message flow	Interconnection Interface SIP (Network B)					
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]					
	faculture dans a serve?					
	[any boundary name]					
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required					
	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.					

Test case number	SS_bcall_036					
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, 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.					
Configuration	Select a proper destination that sends an early ACM in the PSTN/PLMN e.g. announcement					
SIP Parameter	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]					
Message flow						
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →					
	← 100 Trying					
	183 Session Progress(ACM)					
0	Apply post test routine					
Comments	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.					

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						
	Evolit indicator = //EE/(Titto						
	[any boundary name]						
Message flow	[any boundary hamo]						
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 INVITE 100 Trying 180 Ringing(ACM) 200 OK INVITE(ANM) ACK Apply post test routine SIP (Network B) SIP (Network B) APPLIANT APPLIA					
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.					

Test case number	SS_bcall_039						
Test case group	BCALL/successful						
Reference	5.4.3.4, 6.11.2/[24]						
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47						
Test purpose	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.						
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 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 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.						

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 INVITE ← 100 Trying ← 180 Ringing ← 200 OK INVITE ACK Communication ← BYE(REL) 200 OK BYE(RLC) SIP (Network B) SIP (Network B)						
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.</dynamic-pt>					
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)	Interconnection Interface SIP (Network B)					
CASE A	A confirmed session already exists (SDP 1) NVITE(SDP3) → 200 OK INVITE(SDP4) ACK →					
CASE B	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.					
	1					
	Check: Is the codec list consistent with the attribute(s) (bandwidth) regarding					
	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</dynamic-pt>				
Configuration	codecs in table 7.1.2-1 applies.				
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	clock	channels	Supported in	Supported in
			type	rate		network A	network B
VA_01	0	PCMU	Α	8,000	1		
VA_02	3	GSM	Α	8,000	1		
VA_03	4	G723	Α	8,000	1		
VA_04	5	DVI4	Α	8,000	1		
VA_05	6	DVI4	Α	16,000	1		
VA_06	7	LPC	Α	8,000	1		
VA_07	8	PCMA	Α	8,000	1		
VA_08	9	G722	Α	8,000	1		
VA_09	10	L16	Α	44,100	2		
VA_10	11	L16	Α	44,100	1		
VA_13	12	QCELP	Α	8,000	1		
VA_12	13	CN	Α	8,000	1		
VA_13	14	MPA	Α	90,000			
VA_14	15	G728	Α	1 8,000	1		
VA_15	16	DVI4	Α	11,025	1		
VA_16	17	DVI4	Α	22,050	1		
VA_17	18	G729	Α	8,000	1		
VA_18	Dyn	G726-40	Α	8,000	1		
VA_19	Dyn	G726-32	Α	8,000	1		
VA_20	Dyn	G726-24	Α	8,000	1		
VA_21	Dyn	G726-16	Α	8,000	1		
VA_22	Dyn	G729D	Α	8,000	1		
VA_23	Dyn	G729E	Α	8,000	1		
VA_24	Dyn	GSM-EFR	Α	8,000	1		
VA_25	25	CelB	V	90,000			
VA_26	26	JPEG	V	90,000			
VA_27	28	Nv	V	90,000			
VA_28	31	H261	V	90,000			
VA_29	32	MPV	V	90,000			
VA_30	33	MP2T	V	90,000			
VA_31	34	H263	V	90,000			
VA_32	Dyn	H263-1998	V	90,000			
VA_33	Dyn	AMR	Α	8,000	1		
VA_34	Dyn	AMR-WB	Α	16,000	1		
VA_35	Dyn	telephone- event	A	8000	1		

7.1.3 Resource Reservation

T	100 004					
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 Resource reservation successful, segmented status.					
Test purpose	 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 					
Configuration	successfully reserved the QoS in the send and receive directions.					
Configuration SIP Parameter	INIVITE: Cupported: 100rsl presentition					
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 INVITE(SDP1)					
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.					
L						

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	nments 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	l 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.		
Configuration	The session remains unchanged.		
Configuration	NAME OF THE PARTY		
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	[o], [+] and [o]		
Test purpose	Session update requested by the called user is unsuccessful, existing		
rest purpose	session remains unchanged.		
	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		SIP (Network B)
	INVITE ← 180 Ring CANCEL/	ing	
	← 200 OK CANO	EL/BYE	
	← 487 Request To ACK	erminated	
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.		

Tool coop number	66		
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 →		
	→ 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.		
	I topout the tot in foreign and their		

Test case number	SS linelies 011		
	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?		
	epeat this test in reverse direction.		

Reference 6.11.2/[24] SELECTION EXPRESSION 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 intervorking 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 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 INVITE 404 Not Found(REL) ACK SIP (Network B)	Test case number	SS_unsucc_012	
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 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 INVITE 404 Not Found(REL)	Test case group	BCALL/unsuccessful	
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 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 INVITE 404 Not Found(REL)	Reference		
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)	SELECTION EXPRESSION		
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 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 INVITE 404 Not Found(REL)	Test purpose	SIP-I support. Called number is not allocated in the PSTN/PLMN network.	
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 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 INVITE 404 Not Found(REL)			
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 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 INVITE 404 Not Found(REL)			
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 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 INVITE 404 Not Found(REL)			
The called user number is not assigned to the PSTN/PLMN part in Network B 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 INVITE 404 Not Found(REL)			
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 INVITE 404 Not Found(REL)			
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 INVITE 404 Not Found(REL)		· ·	
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 INVITE 404 Not Found(REL)	SIP Parameter		
[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 INVITE 404 Not Found(REL)			
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 INVITE 404 Not Found(REL)		Content-Type: multipart/mixed;boundary=[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 INVITE 404 Not Found(REL)			
Content-Disposition: signal;handling=required REL Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 404 Not Found(REL)			
REL Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 404 Not Found(REL)			
Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 404 Not Found(REL)		Content-Disposition: signar, nandling=required	
Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 404 Not Found(REL)		REI	
[any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE → 404 Not Found(REL)			
Message flow SIP (Network A) Interconnection Interface INVITE		Cause value. 1	
Message flow SIP (Network A) Interconnection Interface INVITE		lany boundary namel	
INVITE → 404 Not Found(REL)	Message flow		
← 404 Not Found(REL)	SIP (Network A)	Interconnection Interface SIP (Network B)	
ACK →			
Comments Establish a communication from network A to Network B, called user number is	Comments		
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		Check: If a Reason header is present, is the cause value equal to the value	
the Cause value of the encapsulated ISUP REL?			
Repeat this test in reverse direction.		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		
	Gadse 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.		

Test case number	SS_unsucc_014
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 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'.
Configuration	
SIP Parameter	480: Reason: Q.850;cause=21 (optional)
Message flow SIP (Network A)	Interconnection Interface INVITE 480 Temporarily Unavailable (REL) ACK SIP (Network B) →
Comments	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.

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		
rest purpose	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 → 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_unsu	rcc 016		
Test case group		nsuccessful		
Reference	7.7.1/[24]			
SELECTION EXPRESSION		A] SE 17 AND SE 47		
Test purpose	SIP-I support. Call clearing due to no answer from the called user initiated			
	Ensure w originatin calling us interwork or BYE to	riginating network. Then the early dialogue is not congressly go network initiate the call clearities is located in the PSTN/PLM ing applies in Network A and the called user. An ISUP RELand the Cause value indicator is	onfirmed by ng after tim N part of Ne ne originatin message is	the called user, the leout of ISUP timer T9 if the letwork A and ISUP - SIP-I leg network sends a CANCEL s encapsulated in the BYE
Configuration				
SIP Parameter	480:			
	[Co	Q.850;cause=19 (optional) ontent-Type: multipart/mixed;bo [any boundary name] ontent-Type: application/isup;veontent-Disposition: signal;hand	ersion=itu-t	92
		REL Cause value: 19		
	[any boundary name]		
Message flow SIP (Network A)	→	Interconnection Interface INVITE 180 Ringing Start timer T9	→	SIP (Network B)
CASE A	+	Timeout T9 CANCEL 200 OK CANCEL 487 Request Terminated ACK	→	
CASE B	(BYE(REL) 200 OK BYE(RLC) 487 Request Terminated ACK	→	
Comments	answer the The ISUF Check: Check: Check: Check: Check: NOTE:	a communication from network ne communication setup. Primer T9 in the PSTN/PLMN end as a CANCEL or BYE request Is a ISUP REL encapsulated it Is the Cause value of the encapt and the Cause value of the encapt as a 487 Request Terminating Are the media streams termin was sent? A ISUP REL is not encapsulated in the setup of the encapsulated is test in reverse direction.	expires is sent by to n a BYE recepture apsulated Figure 15 to send from ated after the	he originating network? quest? REL set to '19'? se value equal to the value in IP REL? the terminating user? ne 200 OK CANCEL/BYE

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	
<u> </u>		
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		
SIF Farailleter			
	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.		
	inchear in is test with an 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-Type: application/isup,version=itu-is2 Content-Disposition: signal;handling=required		
	Content-Disposition: Signal, nanding-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 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		
	'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		
	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		
Message flow	[any boundary name]		
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.		
Configuration SIP Parameter	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. Originating user subscribes to the OIR service		
SIF Parameter	P-Asserted-Identity:		
	Privacy:id OR header OR user		
	From: <sip:anonymous@anonymous.invalid> (optional)</sip:anonymous@anonymous.invalid>		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE +		
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.		

Tost assa number	SS oir C	202		
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.			
	user B is	A and user C are in network B and in network A and is provided with C ted-to user"=Yes.	user C is provided with OIP. The CFU "diverting number is released to	
	permane unconditi diverted-	he served user subscribes Origination to mode), ensure that when user A ional to user C, user C is not informate user receives no identity of the distortion the To header.	calls user B, the call is forwarded	
Configuration	Diverting	user subscribes to the OIR service		
SIP Parameter	INVITE:	no history entry present		
	<sip:< th=""><th>nfo header: userB@networkA?Privacy=history > userC@networkB;cause=302 >;ind</th><th>· · · · · · · · · · · · · · · · · · ·</th></sip:<>	nfo header: userB@networkA?Privacy=history > userC@networkB;cause=302 >;ind	· · · · · · · · · · · · · · · · · · ·	
Message flow	'	· ·		
SIP (Network A)		Interconnection Interface	SIP (Network B)	
	←	INVITE		
	С	FU is performed in Network A		
		INVITE →		
		Apply post test routine		
Comments	Check:	No History-Info header is received		
	Check:	Is the Privacy value history is esca	aped in the hi-targed-to-uri of the	
	diverting user in Network A?			
	Repeat this test in reverse direction.			
	Repeat t	his 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		
rest purpose	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		
	LANA		
	IAM Calling party number		
	Calling party number Screening indicator Notwork provided or user provided verified and		
	Network provided or user provided, verified and		
	passed Presentation restriction		
	restricted		
	Address signal[any boundary name]		
	[any soundary 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 <i>presentation restricted</i> 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		
Message flow	[any boundary name]		
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_0	001		
Test case group	SIP-SIP/	Service/TIP		
Reference	5.2.6.4/[8	3]		
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 INVITE	→	SIP (Network B)
CASE A	←	180 Ringing		
CASE B	←	183 Session Progress		
CASE C	← 20	00 OK INVITE(P-Asserted-Identity) Apply post test routine		
Comments		Is the P-Asserted-Identity is present Progress or in a 200 OK INVITE? his test in reverse direction. The test with all relevant end devices		80 Ringing or 183 Session

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)		
	← 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.		
<u> </u>	repeat the test that an officer office 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)		
	← 200 OK INVITE(ANM) ACK →		
	Apply post test routine		
Comments	Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final		
Comments	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.		
<u> </u>	1 1 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2		

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			
Configuration	'allowed'. 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			
Message flow SIP (Network A)	[any boundary name] Interconnection Interface INVITE(IAM) SIP (Network B)			
	 ← 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 provide not verified'? Check: Is the Address presentation restriction indicator in the Generic number.			
	Check: Is the Address presentation restriction indicator in the Generic number set to 'allowed'? Repeat this test in reverse direction.	₽I		
	proposit this test in reverse direction.			

7.1.5.4 Test purposes for TIR

Test case number	SS tir O	n1		
	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 term	inating user subscribes to the 'TIR'	service	
SIP Parameter	18x/200 OK INVITE			
		P-Asserted-Identity:		
		Privacy: id		
Message flow SIP (Network A)		Interconnection Interface INVITE	→	SIP (Network B)
CASE A	←	180 Ringing		
CASE B	←	183 Session Progress		
CASE C	← 20	00 OK INVITE(P-Asserted-Identity) Apply post test routine		
Comments	Check:	Is the P-Asserted-Identity is prese	ent in the	provisional (18x) or final
		(200 OK) response?		
	Check:	Is the Privacy header in the provis	sional (18	3x) or final (200 OK)
		response is set to 'id'?		
	Repeat th	nis test in reverse direction.		
	Repeat th	nis test with all chosen end devices	S.	

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	<u> </u>		
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		
Message flow SIP (Network A)	[any boundary name] 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/[1	17]		
SELECTION EXPRESSION	SE 24			
Test purpose	Hold the	Hold the session the media stream was previously set to sendrecv.		
	UPDATE attribute	nat the UE A requesting hold of the standard to hold the session. Hold is "a=sendonly". The UE A after reque response containing the SDP with the	done containing the SDP with the sting the hold session receives 200	
Configuration				
SIP Parameter				
Message flow SIP (Network A)	A c	Interconnection Interface onfirmed session already exists	SIP (Network B)	
CASE A	←	INVITE(<mark>sendonly</mark>) 200 OK INVITE (recvonly) ACK	→ →	
CASE B	←	UPDATE(<mark>sendonly</mark>) 200 OK UPDATE (recvonly) Apply post test routine	→	
Comments	Check:		the session on hold by sending an ne version parameter in the SDP 'o'	

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 p	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 <i>receives</i> 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 co	onfirmed session already exists			
CASE A	←	INVITE (<mark>sendonly</mark>)			
		200 OK INVITE (recvonly)	→		
	←	ACK			
		INVITE (<mark>inactive</mark>)	→		
	←	200 OK INVITE (inactive)			
		ACK	→		
CASE B	←	INVITE (<mark>sendonly</mark>)			
		200 OK INVITE (recvonly)	→		
	←	ACK	_		
	_	UPDATE(<mark>inactive</mark>)	→		
	←	200 OK UPDATE (inactive)			
0405.0		LIDDATE (
CASE C	←	UPDATE (<mark>sendonly</mark>)			
		200 OK UPDATE (recvonly)	→ →		
	←	INVITE (inactive)	7		
	~	200 OK INVITE (inactive) ACK	→		
		ACK	7		
CASE D	←	UPDATE (<mark>sendonly</mark>)			
CASE D	•	200 OK UPDATE (recvonly)	→		
		UPDATE (inactive)	→		
	←	200 OK UPDATE (inactive)	7		
		Apply post test routine			
Comments	Check:		et the session on hold by sending an		
	Sileck.		the version parameter in the SDP 'o'		
		line is incremented?			
	Check:		et the session on hold by sending an		
	0.100K.		the version parameter in the SDP 'o'		
		line is incremented?	To to to to paramotor in the obt		
	Repeat th	nis test in reverse direction.			
	1 topout ii	no toot in rovoroo unconon.			

Test case number	SS_hold	003		
Test case group	SIP-SIP/Service/HOLD			
Reference	4.5.2.1/[1			
SELECTION EXPRESSION	SE 24	1		
Test purpose	_	the session the media stream wa	s previously set to sendonly.	
	The second secon			
	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 v	with the attribute "a=sendrecv in the	SDP. The UE A after requesting to	
	resume t	he held session receives 200 OK fin	al response and optionally the	
	attribute	<mark>"a=sendrecv</mark> in the SDP. The a=sen	drecv 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 c	onfirmed session already exists		
CASE A		INVITE (<mark>sendonly</mark>)	→	
	←	200 OK INVITE (recvonly)	_	
		ACK	→	
	•	INVITE (sendrecv)	→	
	←	200 OK INVITE (sendrecv)	•	
		ACK	→	
CASE B		INVITE (<mark>sendonly</mark>)	→	
CASE B	←	200 OK INVITE (recvonly)	7	
		ACK	→	
		UPDATE (<mark>sendrecv</mark>)	÷	
	←	200 OK UPDATE (sendrecv)	•	
		(coa.co.)		
CASE C		UPDATE (<mark>sendonly</mark>)	→	
	←	200 OK UPDATE (recvonly)		
		INVITE (<mark>sendrecv</mark>)	→	
	←	200 OK INVITE (sendrecv)		
		ACK	→	
			_	
CASE D	_	UPDATE (<mark>sendonly</mark>)	→	
	←	200 OK UPDATE (recvonly)		
	_	UPDATE (sendrecv)	→	
	←	200 OK UPDATE (sendrecv)		
Comments	Check:	Apply post test routine	the session on hold by conding an	
Comments	Check:		the session on hold by sending an ne version parameter in the SDP 'o'	
		line is incremented?	ie version parameter in the 3DP 0	
	Check:	Is the user in network A able to ret	rieve the session by sending an	
	J50.		ne version parameter in the SDP 'o'	
		line is incremented? The absence		
		default value.		
	Repeat t	his test in reverse direction.		
	1			

Test case group Reference	Test case number	SS_hold_	004	
Reference 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 received in the SDP. The UE A after requesting to resume the held session receives 200 OK final response and optionally the attribute assession already exists. CASE A Interconnection Interface A confirmed session already exists INVITE (responshy) 200 OK INVITE (recvonly) ACK INVITE (responshy) CASE B INVITE (sendonly) 200 OK INVITE (sendonly) ACK INVITE (recvonly) ACK INVITE (sendonly) 200 OK INVITE (sendonly) ACK ACK INVITE (sendonly) ACK ACK ACK ACK ACK ACK ACK ACK ACK AC				
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 user B the UE-A sends an INVITE or UPDATE to resume the session with the attribute forecast on the session with the attribute forecast optionally the attribute forecast on the following in the SDP. The UE A after requesting to resume the held session receives 200 OK final response and optionally the attribute forecast of the session attribute forecast of the				
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 "arrecorolly in the SDP. The UE A after requesting to resume the held session receives 200 OK final response and optionally the attribute "arrecorolly in the SDP. The UE A after requesting to resume the held session areceives 200 OK final response and optionally the attribute session already exists. CASE A Interconnection Interface A confirmed session already exists. HINTE (recorolly) A confirmed session already exists. A confirmed session already exists. HINTE (recorolly) A confirmed session already exists. A confirmed session already exists. HINTE (recorolly) A confirmed session already exists. A confirmed session already exists. A confirmed session already exists. HINTE (recorolly) A confirmed session already exists. A confirmed session already. A confir			']	
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 according to the SDP. The UE A after requesting to resume the held session receives 200 OK final response and optionally the attribute according in the SDP. Configuration SIP Parameter Message flow SIP (Network A) CASE A Invite (sendonly) 200 OK INVITE (recvonly) ACK INVITE (inactive) ACK INVITE (inactive) ACK INVITE (sendonly) 200 OK INVITE (sendonly) ACK IPDATE (inactive) ACK IPDATE (inactive) CONC INVITE (sendonly) ACK IPDATE (inactive) ACK INVITE (inactive) ACK INVITE (sendonly) CONC INVITE (sendonly) ACK IPDATE (inactive) ACK IPDATE (inactive) ACK IPDATE (inactive) ACK IPDATE (sendonly) ACK ACK ACK IPDATE (sendonly) ACK ACK ACK ACK ACK ACK ACK AC				
	rest purpose	Resume	the session the media stream was	s previously set to mactive.
		The Core	ion is in the "inactive" state. Engure	that the LIE A is requesting to
resume the session with the attribute **a=revorally* in the SDP. The UE A after requesting to resume the held session are receives 200 OK final response and optionally the attribute **a=sendonly* in the SDP. SIP Parameter Message flow SIP (Network A) CASE A **Invite (recorally)** **Invite (recorally)** **A confirmed session already exists** **Invite or (popular (recorally))** **A confirmed session already exists** **Invite or (popular (recorally))** **A confirmed session already exists** **Invite or (popular (recorally))** **A confirmed session already exists** **Invite or (popular (recorally))** **A confirmed session already exists** **Invite or (popular (recorally))** **A confirmed session already exists** **Invite or (popular (recorally))** **A confirmed session already exists** **Invite or (popul				
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) CASE A Interconnection Interface A confirmed session already exists INVITE (sendonly) 200 OK INVITE (fecvonly) ACK INVITE (inactive) ACK INVITE (recvonly) ACK INVITE (recvonly) ACK INVITE (recvonly) ACK INVITE (recvonly) ACK INVITE (sendonly) 200 OK INVITE (fecvonly) ACK INVITE (fecvonly) ACK INVITE (recvonly) ACK INVITE (sendonly) 200 OK INVITE (sendonly) ACK INVITE (sendonly) CASE D CASE				
Configuration SIP Parameter Message flow SIP (Network A) CASE A CASE A CASE A CASE B CASE B CASE B CASE B CASE C CA				
CASE C CASE C CASE D CASE D				
SIP Parameter Message flow Interconnection Interface SIP (Network B)	Configuration	optionally	the attribute a=sendonly in the SD	۲.
Message flow SIP (Network A) CASE A Interconnection Interface				
SIP (Network A) Interconnection Interface A confirmed session already exists INVITE (sendonly) 200 OK INVITE (inactive) ACK INVITE (inactive) ACK INVITE (inactive) ACK INVITE (sendonly) ACK				
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CASE A - INVITE (recvonly) - ACK - ACK - ACK - ACK - ACK - ACK - UPDATE (recvonly) - ACK - ACK - UPDATE (inactive) - INVITE (recvonly) - ACK - ACK - ACK - ACK - UPDATE (inactive) - INVITE (recvonly) - ACK - A	SIP (Network A)	٨٥٥		SIP (Network B)
CASE D CASE C CASE C CASE C CASE C CASE C CASE C CASE D CASE C CASE C	CASEA			
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CASE D CASE D UPDATE (recvonly) 200 OK UPDATE (sendonly) 200 OK UPDATE (recvonly) UPDATE (inactive) 200 OK UPDATE (inactive) 200 OK UPDATE (inactive) UPDATE (recvonly) UPDATE (recvonly) 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?				→
ACK UPDATE (recvonly) ← 200 OK UPDATE (sendonly) CASE D ← UPDATE (sendonly) 200 OK UPDATE (recvonly) UPDATE (inactive) UPDATE (inactive) UPDATE (recvonly) 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 set 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?		←		
UPDATE (recvonly) ← 200 OK UPDATE (sendonly) CASE D ← UPDATE (sendonly) 200 OK UPDATE (recvonly) UPDATE (inactive) UPDATE (inactive) UPDATE (fecvonly) ← 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?				→
CASE D UPDATE (sendonly) 200 OK UPDATE (recvonly) UPDATE (inactive) UPDATE (inactive) UPDATE (recvonly) ✓ 200 OK UPDATE (inactive) UPDATE (sendonly) Apply post test routine 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?				→
CASE D ← UPDATE (sendonly) 200 OK UPDATE (recvonly) UPDATE(inactive) ← 200 OK UPDATE (inactive) UPDATE (recvonly) ← 200 OK UPDATE (sendonly) Apply post test routine 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?		←		
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?				
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?				
UPDATE (inactive) 200 OK UPDATE (inactive) UPDATE (recvonly) 200 OK UPDATE (sendonly) Apply post test routine 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?	CASE D	←	UPDATE (sendonly)	
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 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?			200 OK UPDATE (recvonly)	→
UPDATE (recvonly) 200 OK UPDATE (sendonly) Apply post test routine 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?			UPDATE(<mark>inactive</mark>)	→
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 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?		←	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? 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?			UPDATE (<mark>recvonly</mark>)	→
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?		←	200 OK UPDATE (<mark>sendonly</mark>)	
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?			Apply post test routine	
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?	Comments	Check:	Is the user in network B able to set	the session on hold by sending an
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?				e version parameter in the SDP 'o'
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?				
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:		
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?			INVITE or UPDATE request and th	e version parameter in the SDP 'o'
INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?				
line is incremented?		Check:		
				e version parameter in the SDP 'o'
Repeat this test in reverse direction				
proposit tillo tost in reverse direction.		Repeat th	is test in reverse direction.	

Test case number	SS_hold_005
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 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".
Configuration	
SIP Parameter	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)
CASE A	A confirmed session already exists ← INVITE(sendonly)
CASE A	← INVITE(sendonly) 200 OK INVITE(recvonly)
	← ACK
CASE B	← UPDATE(sendonly)
	200 OK UPDATE (recvonly) →
0	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?
	Repeat this test in reverse direction.
ļ	

Test case number	SS_hold_006		
Test case group		Service/HOLD	
Reference	4.5.2.1/[1	l7]	
SELECTION EXPRESSION	SE 24	-	
Test purpose	Hold the	session the media stream was p	reviously 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	a= <u>inacti</u>	ve .	
Configuration SIP Parameter			
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
SIF (NetWORK A)	Δ	onfirmed session already exists	SIF (Network B)
CASE A	A 0	INVITE(sendonly)	→
	←	200 OK INVITE (recvonly)	-
		ACK	→
	←	INVITE (inactive)	
		200 OK INVITE (inactive)	→
	←	ACK	
0.055		DD (ITT	
CASE B	-	INVITE(sendonly)	→
	←	200 OK INVITE (recvonly) ACK	→
	←	UPDATE (inactive)	7
	~	200 OK UPDATE (inactive)	→
		200 OR OF BATE (Mactive)	,
CASE C		UPDATE (<mark>sendonly</mark>)	→
	←	200 OK UPDATE (recvonly)	
	←	INVITE (inactive)	
		200 OK INVITE (<mark>inactive</mark>)	→
	←	ACK	
CASE D		LIDDATE (condents)	•
CASE D	←	UPDATE (<mark>sendonly</mark>) 200 OK UPDATE (recvonly)	7
	-	UPDATE (inactive)	
	•	200 OK UPDATE (inactive)	→
		Apply post test routine	•
Comments	Check:		et the session on hold by sending an
		INVITE or UPDATE request?	,
	Check:	Is the user in network B able to se	et the session on hold by sending an
			he version parameter in the SDP 'o'
		line is incremented?	
	Repeat t	his test in reverse direction.	

Test case number	SS_hold_	_007		
Test case group	SIP-SIP/	Service/HOLD		
Reference	4.5.2.1/[1	[7]		
SELECTION EXPRESSION	SE 24	-		
Test purpose	Resume	the session the media stream w	as previously set to recvonly.	
		hat the UE A receives an INVITE of		
	resume the session with user B, the UE-A starts sending media. Resume is done			
			ndrecv". The UE A after receiving the	
		of the session sends 200 OK final		
		ute "a= <mark>sendrecv</mark> ". The a=sendrecv the attribute can be omitted.	attribute is the default value	
Configuration	mereiore	the attribute can be offitted.		
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface	SIP (Network B)	
On (Network A)	Δ c	onfirmed session already exists	Oil (Network B)	
CASE A	←	INVITE (sendonly)		
	-	200 OK INVITE(recvonly)	→	
	←	ACK`		
	←	INVITE(sendrecv)		
		200 OK INVITE(<mark>sendrecv</mark>)	→	
	←	ACK		
040F B		LIDDATE (le ala)		
CASE B	←	UPDATE (sendonly)	→	
	←	200 OK UPDATE (recvonly) UPDATE (sendrecv)	7	
	~	200 OK UPDATE (sendrecv)	→	
		Apply post test routine		
Comments	Check:		et the session on hold by sending an	
	INVITE or UPDATE request and the version parameter in the SDP 'o'			
		line is incremented?		
	Check:			
	INVITE or UPDATE request and the version parameter in the SDP 'o'			
	line is incremented?			
	Repeat th	his test in reverse direction.		

Test case number	SS_hold	008	
Test case group		Service/HOLD	
Reference	4.5.2.1/[1		
		7]	
SELECTION EXPRESSION	SE 24 Resume the session the media stream was previously set to inactive.		
Test purpose	Resume	the session the media stream wa	is previously set to inactive.
		sion is in the "inactive" sta <mark>te. Ens</mark> ure	
		TE request requesting to resume th	
		nding media. Resume is done conta	
	"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=sendre	ecv attribute is the default value ther	refore the attribute can be omitted.
Configuration			
SIP Parameter			
Message flow	•		
SIP (Network A)		Interconnection Interface	SIP (Network B)
,	A co	onfirmed session already exists	,
CASE A		INVITE (sendonly)	→
	←	200 OK INVITE (recvonly)	-
	-	ACK	→
	←	INVITE (<mark>inactive</mark>)	
	•	200 OK INVITE (inactive)	→
	←	ACK	
	-	INVITE (<mark>recvonly</mark>)	
	•	200 OK INVITE (sendonly)	→
	←	ACK	7
	~	ACK	
CASE B		INI\/ITE (condonly)	_
CASE B	,	INVITE (sendonly)	→
	←	200 OK INVITE (recvonly)	
	,	ACK	→
	←	UPDATE (inactive)	
	_	200 OK UPDATE (inactive)	→
	←	UPDATE (recvonly)	_
		200 OK UPDATE (<mark>sendonly</mark>)	→
			_
CASE C	_	UPDATE (sendonly)	→
	(200 OK UPDATE (recvonly)	
	←	INVITE (<mark>inactive</mark>)	_
		200 OK INVITE (inactive)	→
	←	ACK	
	←	INVITE (<mark>recvonly</mark>)	
		200 OK INVITE (<mark>sendonly</mark>)	→
	←	ACK	
CASE D		UPDATE (sendonly)	→
	←	200 OK UPDATE (recvonly)	
	←	UPDATE (inactive)	
		200 OK UPDATE (inactive)	→
	←	UPDATE (<mark>recvonly</mark>)	
		200 OK UPDATE (sendonly)	→
		Apply post test routine	
Comments	Check:		t the session on hold by sending an
			ne version parameter in the SDP 'o'
		line is incremented?	,
	Check:		t the session on hold by sending an
			ne version parameter in the SDP 'o'
		line is incremented?	
	Check:	Is the user in network B able to ret	rieve the session by sending an
	330		ne version parameter in the SDP 'o'
		line is incremented?	.5 .5.6.6.1 paramotor in the ODI O
	Repeat th	nis test in reverse direction.	
	Intehear II	no test in reverse unection.	

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		dia etroam was proviously sot to		
rest purpose	Resume the session on both sides the media stream was previously set to			
	inactive.			
	The Session is in the "inactive" state. Ensure			
	resume the session with user B, the UE-A sta			
	INVITE or UPDATE request to resume the se-			
	"a=sendonly in the SDP. The UE A after requ			
	receives 200 OK final response containing the	e SDP with the attribute		
	"a=recvonly. The UE B after requests to resu	me the session receives 200 OK		
	final response containing the SDP with the att			
	a=sendrecv attribute is the default value there			
Message flow				
SIP (Network A)	Interconnection Interface	SIP (Network B)		
on (notwork /t)	A confirmed session already exists	on (Notwork B)		
CASE A	INVITE(sendonly)	→		
CASE A	← 200 OK INVITE (recvonly)	•		
		_		
	ACK	→		
	← INVITE(inactive)			
	200 OK INVITE (inactive)	→		
	← ACK	_		
	INVITE(<mark>sendonly</mark>)	→		
	← 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)	→		
	200 OR OF DATE (Sendred)	•		
CASE C	UPDATE (sendonly)	_		
CASE C		→		
		_		
	200 OK INVITE (inactive)	→		
	← ACK			
	UPDATE (<mark>sendonly</mark>)	→		
	← 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	•		
	Appry post test routille			

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 th	his 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=send resume the session receives 200 OK final resume the session receives 200 OK final resume tribute "a=sendrecv". The UE B after receive sends 200 OK final response containing the SThe a=sendrecv attribute is the default value omitted.	donly". The UE A after requests to sponse containing the SDP with the ing the Resume of the session SDP with the attribute "a=sendrecy".	
Configuration			
SIP Parameter			
Message flow	•		
SIP (Network A) CASE A	Interconnection Interface A confirmed session already exists ← INVITE(sendonly)	SIP (Network B)	
CASE A	200 OK INVITE (recvonly)	→	
	INVITE(inactive)	→	
	← 200 OK INVITE (inactive) ACK	→	
	← INVITE(sendonly) 200 OK INVITE (recvonly)	→	
	← ACK INVITE(<mark>sendrecv</mark>))	
	← 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)	→	
	← UPDATE (sendonly) 200 OK UPDATE (recvonly) INVITE(sendrecy)	→	
	← 200 OK INVITE (sendrecv) ACK	→	
CASE D	 UPDATE (sendonly) 200 OK UPDATE (recvonly) UPDATE (sendonly) 	→ →	
	 ← 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'
	Check:	line is incremented? Is the user in network A able to set the session on hold by sending an
	Check.	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 th	nis test in reverse direction.

-	The second secon		
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-Type: application/isdp,version=id-is2 Content-Disposition: signal;handling=required		
	Content-Disposition, signal, handling=required		
	CPG		
	Generic notification		
	remote hold		
	Tomoto nota		
	[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 →		
	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]				
	o-condonly				
	a=sendonly				
	[any boundary name]				
	[any boundary name] Content-Type: application/isup;version=itu-t92				
	Content-Type: application/isup,version=itu-ts2 Content-Disposition: signal;handling=required				
	CPG				
	Generic notification				
	remote hold				
	[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				
Comments	Apply post test routine Establish a session from Network A to Network B				
Comments	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				
rest purpose	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. The session of th				
	Terminating user in Network B places the session on hold. Originating user in Network A retrieves the session.				
	 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 remete retrieval				
	remote retrieval[any boundary name]				
Message flow	[any boundary name]				
SIP (Network A)	Interconnection Interface SIP (Network B)				
on (notwork /t)	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)				
	Apply post tost routing				
Comments	Apply post test routine Establish a session from Network A to Network B				
Comments	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 group SIP-SIP/Sen/ce/HOLD SELECTION EXPRESSION Network A) SE 17 AND SE 47 AND SE 54 SPI supprose SIP supprove. 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-1 interworking applies in the Network A: Coriginating 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 B retrieves the session. Varify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A retrieves the session. Varify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A. Configuration INVITE: Content-Type: multipart/mixed:boundary=[any boundary name] —[any boundary name] Content-Disposition: signal:handling=required CPG Generic notification remote hold or remote retrieval —[any boundary name]- Message flow SIP (Network A) Interconnection Interface A confirmed session already exists INVITE(sendonly, CPG hold) → 200 OK INVITE (inactive) → ACK INVITE(sendonly, CPG hold) → 200 OK INVITE (inactive) → ACK INVITE(sendonly) → ACK INVITE(sendonly) → ACK INVITE(sendonly) → ACK APPly post test routine Establish as session from Network A lotwork A places the session on hold. Check: Is the "a' attribute in the SIPP set to 'sandonly'? Check: Is the "a' attribute in the SIPP set to 'sandonly'? Check: Is the "a' attribute in the SIPP set to 'sandonly'? Check: Is the "a' attribute in the SIPP set to 'sandonly'? Check: Is the Version parameter in the SIPP incremented? The user in Network A places the session Check: Is the Version parameter in the SIPP incremented? The user in Network B places the session on hold. Check: Is the Version parameter in the SIPP incremented? The user in Network B places the sessio	Test case number	SS_hold_014			
Reference					
SIPI 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 - SIPI 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 places the session. • Originating user in Network A places the session. • Originating user in Network A retrieves the session. • Originating user in Network A retrieves the session. • Originating user in Network A retrieves the session. • Originating user in Network A retrieves the session. • Originating user in Network A retrieves the session. • Originating user in Network A retrieves the session. • Originating user in Network A retrieves the session. • Originating user in Network A retrieves the session. • Originating user in Network A retrieves the session. • Originating user in Network A retrieves the session on hold. • INVITE: Content-Type: multipart/mixed;boundary=[any boundary name] • (PB) Generic notification remote retrieval • (PC) Generic notification signal;handling=required CPG Generic notification signal;handling=required CPG Generic notification signal;handling=required CPG Generic notification promote retrieval • (PN) • (PN					
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 retrieves the session. Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A: Configuration INVITE: Content-Type: multipart/mixed;boundary=lany boundary name]{any boundary name} Content-Type: application/sup;version=itu-f92 Content-Type: app	SELECTION EXPRESSION				
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 places the session on hold. Terminating user in Network A 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. SIP Parameter Notified	Test purpose	terminating user. Retrieve requested by the terminating user.			
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. INVITE		in the Network A:			
Configuration SIP Parameter Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] [any boundary name] Content-Disposition: signal;handling=required CPG Generic notification remote hold or remote retrieval [any boundary name] [any boundary name]		 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 			
SIP Parameter	Configuration				
Content-Type: application/isupy.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 A confirmed session already exists INVITE(sendorly, CPG hold)					
remote retrieval		Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic notification			
Message flow SIP (Network A) Interconnection Interface A confirmed session already exists INVITE (sendonly), CPC hold) ← 200 OK INVITE (recvonly) ACK INVITE (inactive) 200 OK INVITE (inactive) 200 OK INVITE (inactive) 200 OK INVITE (sendonly) ← ACK INVITE (sendonly) ← ACK INVITE (sendonly) ← ACK INVITE (sendorecv) 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 the 'a' attribute in the SDP set to 'sendonly'? Check: Is the 'a' attribute in the SDP set to 'sendonly'? Check: Is the 'a' attribute in the SDP set to 'inactive'? Check: Is the 'a' attribute in the SDP set to 'inactive'? Check: Is the 'a' attribute in the SDP set to 'recvonly'? Check: Is the 'a' attribute in the SDP incremented? The user in Network B retrieves the session Check: Is the 'a' attribute in the SDP incremented? The user in Network B retrieves the session Check: Is the 'a' attribute 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 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 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP set to 'sendrecv'?		remote retrieval			
SIP (Network A) Interconnection Interface A confirmed session already exists INVITE (sendonly, CPG hold) COO K INVITE (recvonly) ACK ACK INVITE (inactive) 200 OK INVITE (inactive) ACK INVITE (sendonly) ACK APPLY post test routine 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 Cendonly? 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 incremented? The user in Network B places the session on hold Check: Is the 'a' attribute in the SDP incremented? The user in Network B retrieves the session Check: Is the Version parameter in the SDP incremented? The user in Network B retrieves the session 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 CPG encapsulated in the INVITE request? Check: Is a CPG encapsulated in the INVITE request? Check: Is a CPG encapsulated in the INVITE request? Check: Is a CPG encapsulated in the INVITE request? Check: Is a CPG encapsulated in the INVITE request? Check: Is a CPG encapsulated in the INVITE request? Check: Is a CPG encapsulated in the INVITE request? Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP incremented?	Mossago flow	[any boundary name]			
## ACK INVITE (sendrecv, CPG retrieval)	SIP (Network A)	A confirmed session already exists INVITE(sendonly, CPG hold) ← 200 OK INVITE (recvonly) ACK INVITE(inactive) 200 OK INVITE (inactive)			
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 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 'a' attribute in the SDP set to 'recvonly'? Check: Is the 'a' attribute in the SDP incremented? The user in Network B retrieves the session 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 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the Version parameter in the SDP incremented?		200 OK INVITE (sendonly) →			
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 'a' attribute in the SDP set to 'sendrecv'? Check: Is the 'a' attribute in the SDP incremented?		← 200 OK INVITE (sendrecv) ACK →			
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 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 'a' attribute in the SDP set to 'sendrecv'? Check: Is the Version parameter in the SDP incremented?	Comments	1171			
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?	Comments				
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?		Check: Is a CPG encapsulated in the INVITE request? Check: Is a Generic notification parameter present the Notification indicator			
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?		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 '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?		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 'sendrecv'? Check: Is the Version parameter in the SDP incremented?		The user in Network B retrieves the session			
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?		Check: Is the Version parameter in the SDP incremented?			
Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the Version parameter in the SDP incremented?		Check: Is a Generic notification parameter present the Notification indicator			
Repeat this test in reverse direction.		Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the Version parameter in the SDP incremented?			

Test case number	SS_hold_015				
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 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 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				
	← INVITE(sendonly) 200 OK INVITE (recyonly)				
	200 OK INVITE (recvonly) → ← ACK				
	₹ ACK				
	INVITE(<mark>inactive</mark> , CPG <mark>hold</mark>) →				
	← 200 OK INVITE (inactive)				
	ACK →				
	NUMBER OF COMMENTS				
	INVITE(<mark>recvonly</mark> , CPG <mark>retrieval</mark>) → 200 OK INVITE (sendonly)				
	ACK →				
	AON				
	← INVITE(sendrecv)				
	200 OK INVITE (sendrecv) →				
	← ACK				
	Apply post test routine				
Comments	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 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 'sendrecy'?				
	Check: Is the Version parameter in the SDP incremented?				
	Repeat this test in reverse direction.				

Test case number	SS_hold_016				
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 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				
0	INVITE request from the Network A.				
Configuration SIP Parameter	IN OTE				
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	[any boundary name]				
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				
Comments	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 'sendrecy'? Check: Is the Varsian parameter in the SDP incremented?				
	Check: Is the Version parameter in the SDP incremented?				
	Repeat this test in reverse direction.				

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	Commun	nication forwarding uncondition	al, basic r	ules.	
		A and user C are in Network A. T with CFU.	he user B is	s in network B and is	
		nat when user A calls user B, the o	call is forwa	rded unconditional to user	
		active call state, ensure the prope			
Configuration	0	delive can etate, enedie the prope	nty or opoo	011.	
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			
Commonto	Chaala	Apply post test routine	ı		
Comments	Check: CDIV unconditional is successful. Check: In the active call state, ensure the property of speech.				
	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 3	30		
Test purpose	Communication	forwarding uncondition	onal, no noti	fication.
	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 =			
SIP Parameter				
Message flow SIP (Network A)	IN C ← IN 180	connection Interface NVITE(Call-ID A-B) CFU is performed NVITE(Call-ID B-C) Ringing(Call-ID C-B) Ringing(Call-ID B-A)	→	SIP (Network B)
	Appl	y post test routine		
Comments	interco	tification regarding call fo nnection interface. in reverse direction.	orwarding in r	network B is received at the

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:		
	<pre><sip:userb@networkb?privacy=history>;index=1,</sip:userb@networkb?privacy=history></pre>		
	<pre><sip: userc@networka;cause="302</pre"><pre>?Privacy=history>;index=1.1</pre></sip:></pre>		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-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</sip:></sip:userb@networkb>		
Message flow			
SIP (Network A)	Interconnection Interface 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 <sip:="" userc@networka;cause="302">;index=1.1</sip:userb@networkb?privacy=history>		
Message flow	, , , , , , , , , , , , , , , , , , , ,		
SIP (Network A)	Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C) Apply post test routine SIP (Network B) → APPLY NETWORK SIP (Network B)		
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 group Reference 4.5.2.6/[3] 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:user@networkB>;index=1, <sip: userc@networka;cause="302">;index=1.1 Message flow SIP (Network A) Interconnection Interface 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 entry is present in one single History-Info header. Poncet the test is reverse direction.</sip:>	Test case number	SS_cfu_006		
SELECTION EXPRESSION SE 25 AND SE 30 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 INVITE: Request line contains ';cause=302' History-Info header: -sip:userB@networkB>index=1, -sip:userB@networkB;index=1, -sip:userC@networkA;cause=302>;index=1.1 Message flow SIP (Network A) Interconnection Interface 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.	Test case group	SIP-SIP/Service/CFU		
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 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 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.</sip:>	Reference	4.5.2.6/[9]		
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 INVITE: Request line contains ';cause=302' History-Info header: sip: userC@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 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.</sip:>	SELECTION EXPRESSION	SE 25 AND SE 30		
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 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 INVITE(Call-ID A-B) CFU is performed Full Call-ID B-C) Apply post test routine 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.</sip:>	Test purpose	Communication forwarding unconditional, diverted-to user receives the		
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 INVITE(Call-ID A-B) CFU is performed TUTE(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.</sip:>		URI of the served user.		
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 INVITE(Call-ID A-B) CFU is performed TUTE(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.</sip:>				
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 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.</sip:userc@networka;cause=302></sip:userb@networkb>		The user A and user C are in network A. The user B is in network B and is		
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 present 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.</sip:>		provided with CFU "Served user allows the presentation of his/her URI to		
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 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 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.</sip:>				
Subscription options: Served user allows the presentation of his/her URI to diverted-to user = Yes INVITE: Request line contains ';cause=302' History-Info header: <ip><ip><ip><ip><ip><ip><ip><ip><ip><ip></ip></ip></ip></ip></ip></ip></ip></ip></ip></ip>				
Served user allows the presentation of his/her URI to diverted-to user = Yes NVITE: 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 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.</sip:>				
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 </sip:></sip:userb@networkb>	Configuration			
Request line contains ';cause=302' History-Info header: 				
History-Info header:	SIP Parameter			
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				
Sip: userC@networkA;cause=302>;index=1.1 Message flow SIP (Network A) Interconnection Interface SIP (Network B)				
Message flow SIP (Network A) Interconnection Interface 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.				
SIP (Network A) Interconnection Interface 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'? Check: Is the cause parameter in the last entry is set to '302'? The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.		<pre><sip: userc@networka;cause="302">;index=1.1</sip:></pre>		
INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C) Apply post test routine 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.		Interconnection Interfere		
CFU is performed NVITE(Call-ID B-C) Apply post test routine 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.	SIP (Network A)			
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.				
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.				
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.				
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.	Comments			
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.	Comments			
(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.				
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.				
NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.				
each history entry is present in one single History-Info header.				
		·		
Inchest this test in reverse direction.		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</sip:></sip:userb@networkb>		
	181 Being Forwarded History-Info header: <sip:userb@networkb>;index=1, <sip: userc@networka;cause="408">;index=1.1 200 OK INVITE</sip:></sip:userb@networkb>		
	History-Info header: <sip:userb@networkb>;index=1, <sip: userc@networka;cause="486">;index=1.1</sip:></sip:userb@networkb>		
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 ar provided with CFU. Ensure that when user A calls user B, the call is forwarded uncondition C user C is user determined user busy.	
Configuration	<u> </u>	
SIP Parameter		
Message flow		
SIP (Network A)	Interconnection Interface SIP (Netwo	rk 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_00)9		
Test case group		ervice/CFU		
Reference	4.5.2.6/[9]			
SELECTION EXPRESSION	SE 25			
Test purpose	Communi	cation forwarding uncondition	nal, unsucces	sful NDUB.
	Ensure tha	A and user C are in network A. T at when user A calls user B, the r C is network determined user I	call is forward	
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 re	eceiving a 486	Busy Here.
	Repeat this	s 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 SIP Parameter	 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 INVITE: no History-Info header
	181 Being Forwarded no History-Info header
Message flow	
SIP (Network A)	Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C) 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.
	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
lest purpose	options is set to presentation not allowed.
	phono to out to procentation not unonout
	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 <mark>deflected) = no</mark>
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 (<i>Diverted-to user</i>) Call diversion information
	Notification subscription options
	presentation not allowed
	Redirecting reason
	unconditional
	Generic notification
	call is diverting
	odii lo divorting
	[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
	Backward call indicator Called party's status indicator no indication Redirection number Address signal (<i>Diverted-to user</i>) Call diversion information Notification subscription options presentation allowed without redirection number Redirecting reason unconditional Generic notification call is diverting
5	[any boundary name]
	Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) ■ 183 Session Progress 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.

SS_cfu_013 SIP-SIP/Service/CFU 6.5/[24] [Network B] SE 17 AND SE 47 AND SE 55 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.
[Network B] SE 17 AND SE 47 AND SE 55 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
[Network B] SE 17 AND SE 47 AND SE 55 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
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
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
Subscription options:
 Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number
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 (<i>Diverted-to user</i>) Call diversion information Notification subscription options
presentation allowed with redirection number Redirecting reason unconditional Generic notification call is diverting
[any boundary name]
Interconnection Interface 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
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'?

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	
Configuration	ISUP/BICC- SIP-I interworking is applicable in Network A. Subscription options:	
Comiguration	 Connected user subscribed to COLR, Permanent = yes 	
SIP Parameter	200 OK	
on randineter	Content-Type: multipart/mixed;boundary=[any boundary name]	
	[any soundary 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	[any soundary name]	
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)	
•	← 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	
<u> </u>	if ISUP/BICC- SIP-I interworking is applicable in Network A.	
Configuration	Subscription options:	
0.5.5	Connected user subscribed to COLR = no	
SIP Parameter	200 OK	
	Content-Type: multipart/mixed;boundary=[any boundary name]	
	[any hayndary nama]	
	[any boundary name] Content-Type: application/isup;version=itu-t92	
	Content-Type: application/isup,version=itu-is2 Content-Disposition: signal;handling=required	
	Content-Disposition: Signal, nandling-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)	
	← 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	
- Comments	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.	
	rropout 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
rest purpose	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
Message flow	[any boundary name]
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →
	CFU is performed
1	(Call-ID B-C, IAM)
Comments	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.
	Interpretation took in fortal or an oblight

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
Test purpose	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 Redirecting reason unconditional
Message flow SIP (Network A)	[any boundary name] Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFU is performed
1	F 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
- Commonto	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.
	1 strange and an analysis

7.1.5.6.2 Communication Forwarding Busy (CFB)

-	100 (004				
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	The user A and user C are in Network A. The user B is in network B and is				
	provided with CFB.					
	Ensure tl	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.				
		neck: In the active call state, ensure the property of speech.				
	Check:	Is the P-Asserted-Identity preser	nt set to the	identity of the originating		
		user?				
	Repeat t	his 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 th 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.	at			
Configuration	Subscription options:				
	 Originating user receives notification that his communication has been diverted = No 				
SIP Parameter					
Message flow SIP (Network A)	Interconnection Interface 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	ne			
	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 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				
SIP Parameter	diversion notification = No				
SIP Parameter	181 Being Forwarded				
	<pre><sip:userb@networkb?privacy=history&reason=sip;cause=486>;index=1, <sip: userc@networka;cause="486?Privacy=history">;index=1.1</sip:></sip:userb@networkb?privacy=history&reason=sip;cause=486></pre>				
Message flow	CSIP. usero@networkA,cause=400?rnvacy=nistory>,index=1.1				
SIP (Network A)	Interconnection Interface SIP (Network B)				
on (Network A)	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			
	<pre><sip:userb@networkb?reason=sip; cause="486">;index=1,</sip:userb@networkb?reason=sip;></pre>			
	<pre><sip: cause="486" userc@networka;="">; index=1.1</sip:></pre>			
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.			
Comments	 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.			

Test case number	SS cfb 005		
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 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.		
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=486'		
	History-Info header:		
	<pre><sip:userb@networkb?privacy=history&reason=sip;cause=486>;index=1, <sip: userc@networka;cause="486">;index=1.1</sip:></sip:userb@networkb?privacy=history&reason=sip;cause=486></pre>		
Message flow	<sip. usero@networka,cause="400">,index=1.1</sip.>		
SIP (Network A)	Interconnection Interface SIP (Network B)		
On (Network A)	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 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.		

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 Yes		
SIP Parameter	INVITE:		
	Request line contains ';cause=486'		
	History-Info header:		
	<pre><sip:userb@networkb?reason=sip;cause=486>;index=1,</sip:userb@networkb?reason=sip;cause=486></pre>		
	<sip: userc@networka;cause="486">;index=1.1</sip:>		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B) →		
	CFB is performed ★ INVITE(Call-ID B-C)		
	- "····-=(oa :2 2 0)		
Comments	Apply post test routine Check: A History-Info header is received in the INVITE contains the URI of		
Comments	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.		
	I representation and the second and		

Test case number	SS_cfb_007	
Test case group	SIP-SIP/Service/CFB	
Reference	4.5.2.6/[9]	
Test purpose Configuration SIP Parameter	SE 26 AND SE 30 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. 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 INVITE:	
	Request line contains ';cause=486' History-Info header:	
Message flow	Colp. address inclination (, dadde = 1002, index=1.1	
SIP (Network A)	Interconnection Interface 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 C-B) ACK(Call-ID C-B) COMMINVITE(Call-ID C-B) ACK(Call-ID C-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	

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_0	009		
Test case group	SIP-SIP/Service/CFB			
Reference	4.5.2.6/[9]			
SELECTION EXPRESSION	SE 26	<i>7</i> 1		
Test purpose		nication forwarding busy, unsu	acceful N	DUB
rest purpose	Commu	incation forwarding busy, unsuc	Cessiui i	ВОВ.
	The user A and user C are in network A. The user B is in network B and is provided with CFB.			
		hat when user A calls user B, the o	call is forwa	arded 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-ÌD A-B)	→	
Comments	Check:	A 181 Being Forwarded is receive	ved at netv	ork 1 originating access.
	Check:	The dialogue is terminated by re		
	Repeat tl	his test in reverse direction.	3	•

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) INVITE(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	ICC off 011	
	SS_cfb_011 SIP-SIP/Service/CFB	
Test case group Reference		
SELECTION EXPRESSION	[0.5/[24] [Network B] SE 17 AND SE 47 AND SE 55	
	SIP-I support. CFB performed in Network B, Notification subscription	
Test purpose	options is set to presentation not allowed.	
	opilono lo dat to procentament not unionoui	
	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.	
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 haundary nama]	
	[any boundary name] Content-Type: application/isup;version=itu-t92	
	Content-Type: application/isup,version=itu-is2 Content-Disposition: signal;handling=required	
	Content-Disposition: signal, nanding-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]	
Message flow	[arry boundary name]	
SIP (Network A)	Interconnection Interface SIP (Network B)	
on (none, no	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: 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.	
	properting test in reverse unection.	

Tost assa number	CC ofb 042
Test case number	SS_cfb_012
Test case group	SIP-SIP/Service/CFB
Reference SELECTION EXPRESSION	[6.5/[24]
	[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) = <mark>yes</mark> , 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 (<i>Diverted-to user</i>)
	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	[arry bournairy name]
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 the Redirection number present?Check: Is Notification subscription options indicator is set to 'presentation'
	Check: Is the Redirection number present?Check: Is Notification subscription options indicator is set to 'presentation allowed without redirection number'?
	Check: Is the Redirection number present?Check: Is Notification subscription options indicator is set to 'presentation'

Toot agas number	CC att 042
Test case number	SS_cfb_013
Test case group	SIP-SIP/Service/CFB
Reference SELECTION EXPRESSION	[6.5/[24]
	[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)
Comments	Apply post test routine
Comments	
	Originating user in Network A establishes a call to user in Network B. Network B
	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A
	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.
	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?
	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'?
	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?
	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?
	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

Test case group	SS_cfb_014		
Reference	SIP-SIP/Service/CFB		
1.10.0.0101100	5.7/[24]		
	Network A] SE 17 AND SE 47 AND SE 53		
	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		
	SUP/BICC- SIP-I interworking is applicable in Network A.		
Configuration	Subscription options:		
OID Danamatan	Connected user subscribed to COLR, Permanent = yes		
SIP Parameter	<pre>200 OK</pre>		
	contone type. Hamparethixed, sealidary—[arry searidary hame]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	ANM		
	Redirection number restriction Presentation restricted		
	Fresentation 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		
l II	Check: Is a 200 OK INVITE received at the interconnection interface?		
	Check: Is an ANM encapsulated in the 200 OK?		
II II	Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation		
	restricted'?		
 F	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		
<u> </u>	if ISUP/BICC- SIP-I interworking is applicable in Network A.		
Configuration	Subscription options:		
OID D	Connected user subscribed to COLR = no		
SIP Parameter	200 OK		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any hayndary nama]		
	[any boundary name] Content-Type: application/isup;version=itu-t92		
	Content-Type: application/isup,version=itu-is2 Content-Disposition: signal;handling=required		
	Content-Disposition: Signal, nanding-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)		
	← 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		
- Commonto	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.		
	rropout 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
rest purpose	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:
3	 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 (<i>Diverting user</i>)
	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
	Oddi Dady
Magaza flavu	[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		
CID Developed	diverting numberinformation		
SIP Parameter	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 (<i>Diverting user</i>)		
	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		
	Cool. 200y		
	[any boundary name]		
Message flow SIP (Network A)	Interconnection Interface INVITE(Call-ID A-B) SIP (Network 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_	-		
Test case group	SIP-SIP/	Service/CFNR		
Reference	4.5.2.6/[9	9]		
SELECTION EXPRESSION	SE 27			
Test purpose	Commu	nication forwarding no reply, ba	sic rules.	
		A and user C are in Network A. T	he user B is	s in network B and is
	T T T T T T T T T T	with CFNR.		
		nat when user A calls user B, the o		rded no reply to user C. In
	the active	e 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	→	
		• • • • • • • • • • • • • • • • • • • •		
Comments	Check:	Apply post test routine CDIV no reply is successful.		
Comments	Check:	In the active call state, ensure the	o proporty	of appeals
	Check:	Is the P-Asserted-Identity preser		
	CHECK.	user?	11 361 10 1116	identity of the originating
	Reneat ti	nis test in reverse direction.		
	I vehear r	iis test iii levelse ullectioll.		

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:		
Comiguration	 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:userb@networkb?privacy=history>		
	<sip: userc@networka;cause="408?Privacy=history">;index=1.1</sip:>		
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			
SIF Farameter	181 Being Forwarded <sip:userb@networkb>;index=1, <sip: userc@networka;cause="408">;index=1.1</sip:></sip:userb@networkb>		
Message flow SIP (Network A)	Interconnection Interface INVITE(Call-ID A-B) ← 180 Ringing(Call-ID B-A) CFB is performed SIP (Network B) →		
	← 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</sip:></sip:userb@networkb?privacy=history>	
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:	
oomiguidion	 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: ;index=1"><sip:userb@networkb>;index=1</sip:userb@networkb> , <sip: userc@network1;cause="408">;index=1.1</sip:>	
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		

Toot coop number	00 star 007		
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:		
	<pre><sip:userb@networkb&reason=sip;cause=408>;index=1,</sip:userb@networkb&reason=sip;cause=408></pre>		
	<sip: userc@networka;cause="486">;index=1.1</sip:>		
	101p. 40010 @110tW01W1,04400=1002,11140X=1.11		
	181 Being Forwarded		
	History-Info header:		
	<sip:userb@network>;index=1,</sip:userb@network>		
	<sip: userc@networka;cause="408">;index=1.1</sip:>		
	Soprassis Sistematical formation and the second sistematical formation		
	200 OK INVITE		
	History-Info header:		
	<sip:userb@networkb>;index=1,</sip:userb@networkb>		
	<sip: userc@networka;cause="408">;index=1.1</sip:>		
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'?		
	TOTAL CONTROL OF THE SECTION AND CONTROL OF THE		
	Check: Is the cause parameter in the last entry is 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		
	Check: Is the cause parameter in the last entry is set to '408'?		

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 U	DUB.	
	The user A and user C are in network A. The user B is in provided with CFNR. Ensure that when user A calls user B, the call is forwards and user C is user determined user busy.		
Configuration	•		
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface INVITE(Call-ID A-B) →	SIP (Network 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 Repeat this test in reverse direction.	Busy Here.	
	Repeat this test in reverse direction.		

Test case number	SS ofnr	000		
		SS_cfnr_009		
Test case group	_	SIP-SIP/Service/CFNR		
Reference	4.5.2.6/[9)]		
SELECTION EXPRESSION	SE 27			
Test purpose	Commu	Communication forwarding no reply, unsuccessful NDUB.		
	provided	A and user C are in network A. Thwith 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 re	ceiving a	186 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		
	SIP-SIP/Service/CFNR		
Test case group Reference			
SELECTION EXPRESSION	[6.5/[24] [Network B] SE 17 AND SE 47 AND SE 55		
	SIP-I support. CFNR performed in Network B, Notification subscription		
Test purpose	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 (<i>Diverted-to user</i>)		
	Call diversion information		
	Notification subscription options		
	presentation not allowed		
	Redirecting reason		
	No reply		
	Generic notification		
	call is diverting		
	[any boundary name]		
	Interconnection Interface INVITE(Call-ID A-B) 180 Ringing (Call-ID B-A, ACM) CFNR is performed INVITE(Call-ID B-C, IAM) 183 Session Progress Apply post test routine SIP (Network B) → 180 (Network B) → 180 (Network B)		
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.		

Toot coop number	100 of the 040
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:
Comiguration	 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
	Event indicator Alerting or Progress Redirection number Address signal (<i>Diverted-to user</i>) 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]	
Reference 6.5/[24]	
SELECTION EXPRESSION [Network B] SE 17 AND SE 47 AND SE 55	
L 1	ation subscription
Test purpose SIP-I support. CFNR performed in Network B, Notifical options is set to presentation allowed with redirection	
The user A and user C are in Network A. The user B is in of Network B and is provided with CFNR, Calling user rechis call has been diverted (forwarded or deflected) = yes, number.	ceives notification that
Ensure that when user A calls user B, the call is forwarde user A is notified of call diversion and informed of the diversion and informed of the diversion information is present in the encapsulated Redirection number and Call diversion information if SIP-interworking is applicable in Network B.	erted-to number. d CPG contained in the
Configuration Subscription options:	
Calling user receives notification that his call has been seen as a contract of the contr	en diverted (forwarded
or deflected) = yes, with diverted-to user number	
SIP Parameter 183 Session Progress	
Content-Type: multipart/mixed;boundary=[any bou	undary 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 (<i>Diverted-to user</i>) Call diversion information	
Notification subscription options	
presentation allowed with redirection	n number
Redirecting reason	ii fidifibei
No reply	
Generic notification	
call is diverting	
[any boundary name] Message flow	
SIP (Network A) Interconnection Interface 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	SIP (Network B)
Comments Originating user in Network A establishes a call to user in	Network R Network P
performs the diversion to a user in Network A	I NELWOIK D. NELWOIK B
Check: 183 Session Progress is received at the interco	onnection interface
Check: Is an CPG encapsulated in the 183?	micolion interiace.
Check: Is the Called party's status indicator set to 'no in	ndication'?
Check: Is the Redirection number present?	
Check: Is Notification subscription options indicator is s	set to 'presentation
	- 1
allowed with redirection number'?	
Check: Is the Redirecting reason set to 'No reply'?	

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.	:- f			
	Ensure that when user A calls user B, the call a Redirection number restriction parameter is				
	restricted in the encapsulated ANM contained				
	ISUP/BICC- SIP-I interworking is applicable in				
Configuration	Subscription options:	THE WORA.			
Comiguration	 Connected user subscribed to COLR, Pe 	rmanent – ves			
SIP Parameter	200 OK	imanem – yes			
on raidmotor	Content-Type: multipart/mixed;bounda	rv=[anv boundarv name]			
	,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,	, [,,,			
	[any boundary name]				
	Content-Type: application/isup;version				
	Content-Disposition: signal;handling=re	equired			

	ANM				
	Redirection number restriction Presentation restricted				
	Presentation restricted				
	[any boundary name]				
Message flow					
SIP (Network A)	Interconnection Interface	SIP (Network B)			
	INVITE(Call-ID A-B), IAM	→			
•					
_	CFNR is performed				
•	=(0a 2 0)	_			
	180 Ringing (Call-ID C-B, ACM)	→			
•		→			
•	200 OK INVITE (Call-ID C-B, ANM) ACK (Call-ID B-C)	7			
•					
	ACK (Call-ID A-B)	→			
	Apply post test routine	•			
Comments	Originating user in Network A establishes a ca	all to user in Network B. Network B			
	performs the diversion to a user in Network A	15 Coo. III Hotholik B. Hotholik B			
	Check: Is a 200 OK INVITE received at the	interconnection interface?			
	Check: Is an ANM encapsulated in the 200				
	Check: Is the ISUP/BICC Redirection numb				
	restricted'?				

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]		
	familian daminana		
	[any boundary name] Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	Content-Disposition. Signal, nanuling=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				
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:			
Comiguration	 Served user releases his/her number to diverted-to user = Release diverting 			
	number information			
SIP Parameter	INVITE			
on ranamotor	Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name] Content-Type: application/isup;version=itu-t92			
	Content-Type: application/isup,version=itd-ts2 Content-Disposition: signal;handling=required			
	IAM			
	Redirecting number			
	Address presentation restricted indicator			
	presentation allowed			
	Address signal (<i>Diverting user</i>) 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			
Message flow	[any boundary name]			
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 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			
Test purpose	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:			
Comiguration	 Served user releases his/her number to diverted-to user = Do not release 			
	diverting numberinformation			
SIP Parameter	INVITE			
on rarameter	Content-Type: multipart/mixed;boundary=[any boundary name]			
	Contone Typo: manaparemixod, soundary – [arry soundary harno]			
	[any boundary name]			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	<u>IAM</u>			
	Redirecting number			
	Address presentation restricted indicator			
	presentation restricted			
	Address signal (<i>Diverting user</i>)			
	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			
Manager diam	[any boundary name]			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)			
SIF (NetWORK A)	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.			
<u> </u>	Last and the second supplies			

7.1.5.6.4 Communication Forwarding Not Logged in (CFNL)

-	-1			
Test case number	SS_cfnl_001			
Test case group	SIP-SIP/Service/CFNL			
Reference	4.5.2.6/[9	9]		
SELECTION EXPRESSION	SE 28			
Test purpose	Commu	nication forwarding not logged i	in, basic ru	iles.
	provided	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		
		active call state, ensure the prope		
Configuration	O. III IIIE	donvo dan state, ensure the prope	nty or spec	OII.
SIP Parameter				
Message flow				
SIP (Network A)	←	Interconnection Interface INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C)	→	SIP (Network B)
	+	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		The CDIV not logged in is succe		
	Check:	In the active call state, ensure th		
	Check:	Is the P-Asserted-Identity preser user?	nt set to the	eldentity of the originating
	Repeat tl	his test in reverse direction.		

Test case number	SS_cfnl_00	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.			
	provided wi that his con Ensure that	and user C are in Network A. th CFNL, subscription option: nmunication has been diverted when user A calls user B, theng user is not notified.	Originating to	user receives notification
Configuration	Subscription options:			
	 Origina diverte 	ating user receives notification d = No	that his com	nmunication has been
SIP Parameter	<u> </u>	WITOTOM - ITO		
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		lo notification regarding call fonterconnection interface.	rwarding in i	network B is received at
	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:		
3	 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:userb@networkb?privacy=history>		
	<sip: userc@networka;cause="404?Privacy=history">;index=1.1</sip:>		
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:		
Comgulation	 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</sip:userc@networka;cause=404></sip:userb@networkb>		
Message flow SIP (Network A)	Interconnection Interface INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) INVITE(Call-ID B-C) 181 Being Forwarded (Call-ID B-A) 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:			
	<pre><sip:userb@networkb?privacy=history>;index=1,</sip:userb@networkb?privacy=history></pre>			
	<sip: userc@network1;cause="404">;index=1.1</sip:>			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE(Call-ID A-B) →			
	CFNL is performed ★ INVITE(Call-ID B-C)			
	- (
Comments	Apply post test routine Check: A History-Info header is received in the INVITE contains the URI of			
Comments	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.			
1	- 			

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 CONTRACTOR OF THE PROPERTY OF THE PROPE		
	Request line contains ';cause=404'		
	History-Info header:		
	<sip:userb@networkb>;index=1,</sip:userb@networkb>		
	<sip: userc@networka;cause="404">;index=1.1</sip:>		
Message flow	OID (No. 1. D)		
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B) →		
	CFNL is performed ★ INVITE(Call-ID B-C)		
	← INVITE(Call-ID B-C) Apply post test routine		
Comments	Check: A History-Info header is received in the INVITE contains the URI of		
Comments	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.
Configuration	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. 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: <a header="" history="" href="mailto:sip:userB@networkB&Reason=B</th></tr><tr><th></th><th>194 Paing Enwarded</th></tr><tr><th></th><th>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</th></tr><tr><th>Message flow</th><th>Csip. dsero enetwork, cadse=+0+2, index=1.1</th></tr><tr><th>ŠIP (Network A)</th><th>Interconnection Interface INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) I81 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) ACK(Call-ID A-B) Apply post test routine</th></tr><tr><th>Comments</th><th>Check: A History-Info header is received in the INVITE at the interconnection</th></tr><tr><th></th><th>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 " info="" one"="" or<="" th="">
	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.	
rest purpose	Communication forwarding not logged in, unsuccessful obob.	
	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 use	\r
	C and user C is user determined user busy.	1
Configuration	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-ÌD 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.	
	repeat this test in reverse direction.	

SS_cfnl_009
4.5.2.6/[9]
ES 183 004
SE 28
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.
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) →
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	 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 INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) 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.
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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
	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)
	F 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.

Toot agas number	CC -f-1 040
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) = ves, 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 (<i>Diverted-to user</i>)
	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 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.
	in topoda and took in rovered 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:		
oomigaranon	 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-Type: application/isup,version=itu-toz 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)		
Comments	Apply post test routine 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:		
OID D	• Connected user subscribed to COLR, Permanent = yes		
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any houndary name]		
	[any boundary name] Content-Type: application/isup;version=itu-t92		
	Content-Type: application/isup,version=itu-t92 Content-Disposition: signal;handling=required		
	Goriterit-Disposition. Signat, handling-required		
	ANM		
	Redirection number restriction		
	Presentation restricted		
	[any boundary name]		
Message flow	Interconnection Interface CID (Network D)		
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B), IAM → CFNL is performed		
	F 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			
reet pan peee	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 D. II. di			
	Redirection number restriction not present			
	[any houndary name]			
Message flow	[any boundary name]			
SIP (Network A)	Interconnection Interface SIP (Network B)			
on (Notwork A)	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.			

Test case number	SS_cfnl_016		
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		
rest purpose	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.		
Configuration	Subscription options:		
Comiguration	 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 (<i>Diverting user</i>) 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]		
Message flow	International Control		
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B) →		
	CFNL is performed NVITE(Call-ID B-C. IAM)		
`	· · · · · · · · · · · · · · · · · · ·		
Comments	Apply post test routine Originating user in Network A establishes a call to user in Network B. Network B		
Comments	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.		

Test case number	SS_cfnl_017		
Test case group	SIP-SIP/Service/CFNL		
Reference	7.1/[24]		
SELECTION EXPRESSION			
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		
OID Developed	diverting numberinformation		
SIP Parameter	INVITE Content Turner resulting at / resisted the sun dam. James he condom up are all		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Type: application/isup,version=itu-is2 Content-Disposition: signal;handling=required		
	Content-Disposition: signal, nationing-required		
	IAM		
	Redirecting number		
	Address presentation restricted indicator		
	presentation restricted		
	Address signal (<i>Diverting user</i>) 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	[any boundary name]		
SIP (Network A)	Interconnection Interface SIP (Network B)		
on (Notwork A)	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_0	01		
Test case group	SIP-SIP/	Service/CD		
Reference	4.5.2.6/[9	9]		
SELECTION EXPRESSION	SE 29			
Test purpose	Commu	nication deflection during alertir	ng, basic ru	ıles.
	The user	A and user C are in Network A. T	he user B is	s in network B and is
	1-1	with CDa.		
		nat when user A calls user B, the o		•
	C. In the	active call state, ensure the prope	rty of speed	ch.
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	→	
		• • • • • • • • • • • • • • • • • • • •		
Comments	Check:	Apply post test routine CDa is successful.		
Comments	Check:	In the active call state, ensure th	o proporty	of appeals
	Check:	Is the P-Asserted-Identity preser		
	CHECK.	user?	ii sei io ille	identity of the originating
	Reneat th	nis test in reverse direction.		
	I vehear r	iis test iii ieveise ulieutiuli.		

Test case number	SS_cd_0	nn2		
Test case group		SIP-SIP/Service/CD		
Reference	4.5.2.6/[9			
SELECTION EXPRESSION	SE 29	·J		
Test purpose		nication deflection immediate, b	asic rules.	
		, .		
	Ensure th	nat when user A calls user B which	n deflects th	e communication towards
	user C im	nmediately (i.e. before alerting star	rts), the call	is forwarded to user C. In
	the active	e call state, ensure the property of	speech.	
Configuration				
SIP Parameter				
Message flow		Interconnection Interface		SID (Notwork D)
SIP (Network A)			→	SIP (Network B)
		INVITE(Call-ID A-B) CDi is performed	7	
	←	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-ÌD A-B)	→	
		Communication		
		Apply post test routine		
Comments	Check:	02:10 04:00001411		
	Check:	In the active call state, ensure th	e property o	f speech.
	Check:	Is the P-Asserted-Identity preser	nt set to the	identity of the originating
		user?		
	Repeat th	nis 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		
rest purpose	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:		
	<pre><sip:userb@networkb?privacy=history&reason=sip;cause=302>;index=1,</sip:userb@networkb?privacy=history&reason=sip;cause=302></pre>		
Manager (Inc.)	<sip: userc@networka;cause="480?Privacy=history">;index=1.1</sip:>		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)		
SIF (Network A)	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</sip:></sip:userb@networkb?reason=sip;cause=302>		
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		
root purpode	receive the URI of the served user.		
	Toolife the office and solved about		
	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:		
	<pre><sip:userb@networkb?privacy=history&reason=sip;cause=302>;index=1,</sip:userb@networkb?privacy=history&reason=sip;cause=302></pre>		
	<sip: userc@networka;cause="480">;index=1.1</sip:>		
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</sip:></sip:userb@networkb?reason=sip;cause=302>		
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_0	09		
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. T	he user B i	s 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 re	eceiving a 4	186 Busy Here.
	Repeat th	nis test in reverse direction		

Toot coop number	SS ad 010		
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)		
Comments	Apply post test routine Check: No History-Info header is received in the INVITE at the interconnection		
Comments	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		
	SIP-SIP/Service/CD		
Test case group Reference			
SELECTION EXPRESSION	[6.5/[24] [Network B] SE 17 AND SE 47 AND SE 55		
	SIP-I support. CD performed in Network B, Notification subscription		
Test purpose	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:		
Joinigaranon	Calling user receives notification that his call has been diverted (forwarded)		
	or deflected) = no		
SIP Parameter	183 /180		
<u> </u>	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 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.		
	1. (apact time toot in foreign amounting		

Test case number	SS cd 012		
	SIP-SIP/Service/CD		
Test case group Reference			
SELECTION EXPRESSION	[6.5/[24]		
	[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:		
301111garation	 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		
	 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		
Comments	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		
rest purpose	options is set to presentation allowed with redirection number.		
	opiniono lo cost to procentament anomou manifestira		
	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]		
	facult and demonstrated		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	Content-Disposition. Signal, nanding=required		
	ACM/CPG		
	Redirection number		
	Address signal (<i>Diverted-to user</i>)		
	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	Interconnection Interfere		
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-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 purpose SIP-I support. Onumber. The user A and of Network B and the COLR service Ensure that whee Redirection number. Configuration Subscription of	AND SE 47 AND SE 53 Deformed in Network B, Restriction of the Redirection user C are in Network A. The user B is in the PSTN/PLMN part d is provided with CDi or CDa, Diverted-to user is subscribed to se in Permanent mode. n user A calls user B, the call is deflected to user C, a ber restriction parameter is present set to 'Presentation encapsulated ANM contained in the 200 OK INVITE if -I interworking is applicable in Network A.
SELECTION EXPRESSION [Network A] SE Test purpose SIP-I support. Onumber. The user A and of Network B and the COLR service Ensure that when Redirection numbers restricted in the ISUP/BICC- SIP Configuration Subscription of	user C are in Network A. The user B is in the PSTN/PLMN part d is provided with CDi or CDa, Diverted-to user is subscribed to se in Permanent mode. n user A calls user B, the call is deflected to user C, a ber restriction parameter is present set to 'Presentation encapsulated ANM contained in the 200 OK INVITE if
Test purpose SIP-I support. Conumber. The user A and of Network B and the COLR service Ensure that when Redirection number. Test purpose The user A and of Network B and the COLR service Ensure that when Redirection number. Suppose Suppose Subscription of Subscription of Subscription of Suppose Su	user C are in Network A. The user B is in the PSTN/PLMN part d is provided with CDi or CDa, Diverted-to user is subscribed to se in Permanent mode. n user A calls user B, the call is deflected to user C, a ber restriction parameter is present set to 'Presentation encapsulated ANM contained in the 200 OK INVITE if
Test purpose SIP-I support. Conumber. The user A and of Network B and the COLR service Ensure that when Redirection number. Test purpose The user A and of Network B and the COLR service Ensure that when Redirection number. Suppose Suppose Subscription of Subscription of Subscription of Suppose Su	user C are in Network A. The user B is in the PSTN/PLMN part d is provided with CDi or CDa, Diverted-to user is subscribed to se in Permanent mode. n user A calls user B, the call is deflected to user C, a ber restriction parameter is present set to 'Presentation encapsulated ANM contained in the 200 OK INVITE if
number. The user A and of Network B and the COLR service Ensure that when Redirection num restricted' in the ISUP/BICC- SIP Configuration Subscription of	user C are in Network A. The user B is in the PSTN/PLMN part d is provided with CDi or CDa, Diverted-to user is subscribed to see in Permanent mode. n user A calls user B, the call is deflected to user C, a ber restriction parameter is present set to 'Presentation encapsulated ANM contained in the 200 OK INVITE if
of Network B and the COLR service Ensure that when Redirection number restricted in the ISUP/BICC- SIP Configuration Of Network B and the College And the Co	d is provided with CDi or CDa, Diverted-to user is subscribed to be in Permanent mode. n user A calls user B, the call is deflected to user C, a ber restriction parameter is present set to 'Presentation encapsulated ANM contained in the 200 OK INVITE if
of Network B and the COLR service Ensure that when Redirection number restricted in the ISUP/BICC- SIP Configuration Of Network B and the College And The Co	d is provided with CDi or CDa, Diverted-to user is subscribed to be in Permanent mode. n user A calls user B, the call is deflected to user C, a ber restriction parameter is present set to 'Presentation encapsulated ANM contained in the 200 OK INVITE if
the COLR service Ensure that whee Redirection num restricted' in the ISUP/BICC- SIP Configuration Subscription of	e in Permanent mode. n user A calls user B, the call is deflected to user C, a ber restriction parameter is present set to 'Presentation encapsulated ANM contained in the 200 OK INVITE if
Ensure that whe Redirection num restricted' in the ISUP/BICC- SIP Configuration Subscription o	n user A calls user B, the call is deflected to user C, a ber restriction parameter is present set to 'Presentation encapsulated ANM contained in the 200 OK INVITE if
Redirection num restricted' in the ISUP/BICC- SIP Configuration Subscription o	ber restriction parameter is present set to 'Presentation encapsulated ANM contained in the 200 OK INVITE if
restricted' in the ISUP/BICC- SIP Configuration Subscription o	encapsulated ANM contained in the 200 OK INVITE if
ISUP/BICC- SIP Configuration Subscription o	
Configuration Subscription o	-I interworking is applicable in Network A.
	user subscribed to COLR, Permanent = yes
SIP Parameter 200 OK	Type: multipart/mixed:houndary-lany houndary namel
Content-	Type: multipart/mixed;boundary=[any boundary name]
lany ho	undary name]
	undary паттеј Гуре: application/isup;version=itu-t92
	Disposition: signal;handling=required
Contone	Sisposition: dignar, narrating—roquirod
ANM	
R	edirection number restriction
	Presentation restricted
	undary name]
Message flow	
, , , ,	connection Interface SIP (Network B)
	ITE(Call-ID A-B), IAM →
	ng (Call-ID B-A) in case CDa
	CD is performed
	NVITE(Call-ID B-C)
	nging (Call-ID C-B, ACM) → Ringing (Call-ID B-A)
	NVITE (Call-ID C-B, ANM)
	ACK (Call-ID B-C)
	OK INVITE (Call-ID B-A)
	ACK (Call-ID A-B) →
	ly post test routine
	in Network A establishes a call to user in Network B. Network B
	ersion to a user in Network A
	00 OK INVITE received at the interconnection interface?
	ANM encapsulated in the 200 OK?
	ISUP/BICC Redirection number restriction set to 'Presentation
restric	
Repeat this test	** * ·

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		
reet par peee	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		
	ANIM		
	ANM		
	Redirection number restriction		
	Presentation allowed		
	or Redirection number restriction not present		
	Redirection humber restriction not present		
	[any boundary name]		
Message flow	[arry boundary name]		
SIP (Network A)	Interconnection Interface SIP (Network B)		
, (INVITE(Call-ID A-B), IAM →		
	← 180 Ringing (Call-ID B-A) in case CDa		
	ČD 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 common A and down an O and in Nationals A. The common B in its the DOTAL/DLAMA and		
	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 haundary name]		
	[any boundary name] Content-Type: application/isup;version=itu-t92		
	Content-Type: application/isup,version=itu-ts2		
	Content Disposition: Signal, nama ing-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 haymdon, none]		
Message flow	[any boundary name]		
SIP (Network A)	Interconnection Interface SIP (Network B)		
on (notwork //)	INVITE(Call-ID A-B) →		
	← 180 Ringing (Call-ID B-A) in case CDa		
	ČD 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.		
	Tireties and recommended amounting		

Test case number	SS_cd_017		
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 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.		
Configuration	Subscription options:		
	Served user releases his/her number to diverted-to user = Do not release		
OLD D	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 Dedinaction rumber		
	Redirecting number Address presentation restricted indicator presentation restricted		
	Address signal (<i>Diverting user</i>)		
	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]		
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 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 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.		
	1. repeat and took in reverse anothern		

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 conference="" focus;method="INVITE" of=""> 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 conference="" focus;method="INVITE" of=""> NOTIFY(B2, 100) Content-Type: message/sipfrag SIP/2.0 100 INVITE: Request URI: uri of conference focus</uri></uri>	
	From: user B2 NOTIFY(B2, 200) Content-Type: message/sipfrag SIP/2.0 200 OK	

Message flow SIP (Network /	Δ)	Interconnection Interface	SIP (Network B)
		to user B1 from Network A to N	
		to user B2 from Network A to N	
Lotabilon a V		A establishes a 3PTY conversat	
	333.	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			fter the confirmed communication to
		and B2 are set on hold	
	Check:		ER method sent to user B1 and B2
			ice focus and is the method parame
		set to 'INVITE'.	
	Check:	The NOTIFY after the REFER re	equest contains the 'SIP/2.0 100'
		message body.	
	Check:		ser B1 and user B2 to the conference
		focus the Request URI is used for	rom the Refer-To header of the
		received REFER request.	
	Check:		equest contains the 'SIP/2.0 200 OK
		message body.	
	Check:	The original session is terminate	ed by user A.
	Repeat t	his 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 relNVITE performed by the AS in network A.			
rost parposs	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></b1>			
	From: <usera></usera>			
	To: <userb1></userb1>			
	Call-ID: A-B1			
	P-Asserted-Identity: <usera></usera>			
	SDP: a=sendrecv			
	INVITE <b2></b2>			
	From: <usera> Call-ID: A-B2</usera>			
	To: <userb2></userb2>			
	P-Asserted-Identity: <usera></usera>			
	SDP: a=sendrecv			
Message flow	ODI : a=serialecv			
SIP (Network A)	Interconnection Interface SIP (Network B)			
	d session to user B1 from Network A to Network B and put it on hold			
	d 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			
	SIP-SIP/Service/CONF			
Test case group	5.4/[24]			
Reference				
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></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</b2>			
	CPG			
	Generic Notification			
	Conference established			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)			
Establish a confirmed se	ssion from User A in Network A to user B1 in Network B and put it on hold irmed 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			
Comments	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.			

Test case number	SS_conf_004			
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 disconnects one of the remote users.			
rest purpose	on -moor line working. Gerved 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'.			
Configuration	ISUP/BICC interworking applies in Network A			
	User in Network A is subscribed to the 3PTY supplementary service			
SIP Parameter	INFO <b2></b2>			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	CPG			
	Generic Notification			
Manager Slave	Conference disconnected			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)			
	sion from User A in Network A to user B1 in Network B and put it on hold			
	rmed session from User A in Network A to user B2 in Network B			
LStabilish a comi	User A establishes a 3PTY conversation			
	BYE(Call-ID A-B1, REL) →			
	€ 200 INFO			
	- 200 0			
	INFO(Call-ID A-B2, CPG) →			
	← 200 INFO			
	Apply post test routine			
Comments	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.			

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.				
root pai pood	on the or morning contour desired and or any 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></b1>				
	Content-Type: application/isup;version=itu-t92				
	Content-Disposition: signal;handling=required				
	CPG				
	Generic Notification Conference disconnected				
	INFO .DO.				
	INFO <b2></b2>				
	Content-Type: application/isup;version=itu-t92				
	Content-Disposition: signal;handling=required				
	CPG				
	Generic Notification				
	Conference disconnected				
Message flow	Contenence disconnected				
SIP (Network A)	Interconnection Interface SIP (Network B)				
	ssion from User A in Network A to user B1 in Network B and put it on hold				
	firmed session from User A in Network A to user B2 in Network B				
Establish a som	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.				
	- i				

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</b1>			
	CPG			
	Generic Notification			
	conference established			
	INFO2 <b1></b1>			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	CPG Generic Notification			
	Other party added			
	INFO <b2></b2>			
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required			
	CPG			
	Generic Notification			
	conference established			
Message flow SIP (Network A) Establish a confi	Interconnection Interface SIP (Network B) irmed 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 so	ession from User A in Network A to user B2 in Network B and add to the conference			
	INFO2(Call-ID A-B2, CPG)			
	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</b1>		
	INFO2 <b1> CPG Generic Notification= Other party isolated</b1>		
	INFO1 <b2> CPG Generic Notification= reattached</b2>		
	INFO2 <b2></b2>		

Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
	a CONF com	munication with User B1 and U	•
	User A isola	tes User B1 from the CONF con	versation
		INFO1(Call-ID A-B1, CPG)	→
	←	200 INFO	
		INFO1(Call-ID A-B2, CPG)	→
	←	200 INFO	
	User A reatta	aches User B1 to the CONF con	versation
		INFO2(Call-ID A-B2, CPG)	→
	←	200 INFO	
		INFO2(Call-ID A-B2, CPG)	→
	←	200 INFO	
		Apply post test routine	
Comments	User A I	nvokes a CONF conversation with	User B1 and User b2 in Network B
	User A s	plits user B1 in Network B from th	ne CONF conversation
	Check:		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 er	ncapsulated CPG?
	User A r	eattaches 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 encapsulate	ed 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 t	his test in reverse direction	

Tost case number	ICC conf 000		
Test case number	SS_conf_008 SIP-SIP/Service/CONF		
Test case group			
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></b1>		
	CPG Generic Notification= conference disconnected		
	INFO2 <b1></b1>		
	CPG		
	Generic Notification=Other party split		
	INFO1 <b2> CPG Generic Notification=Conference established</b2>		
	INFO2 <b2> CPG</b2>		
Manager Starre	Generic Notification= Other party added		
	Interconnection Interface SIP (Network B) CONF communication with User B1 and User B2 in Network B		
Us	ser A isolates User B1 from the CONF conversation		
	INFO1(Call-ID A-B1, CPG) → 200 INFO		
	INFO1(Call-ID A-B2, CPG) →		
	← 200 INFO		
Us	ser 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. Chack: Is an INFO request sent to user B1 and the Generic notification is set		
	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		
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7.1.5.8 Anonymous Communication Rejection (ACR) and Communication Barring (CB)

Test case number	SS_acr-o	cb 001		
Test case group		SIP-SIP/Service/ACR-CB		
Reference		4.5.2.6/[12]		
SELECTION EXPRESSION	SE 32	·- <u>1</u>		
Test purpose		Call Barring performed in network B for user B.		
reet par peee	Jun Jun	g por oou not o 2	, uoo. D .	
	User A is	s located in network A and user E	3 is located	d in network B and is
		ed to the Incoming Call Barring s		
		hat a communication from user A		d in network B by sending a
		line due to the Call Barring service		
Configuration		s subscribed to the incoming Cal		
	list)	3	J	(3
SIP Parameter	INVITE			
		P-Asserted-Identity: <uri of="" th="" us<=""><th>ser A></th><th></th></uri>	ser A>	
Message flow		-		
SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE	→	
	←	603 (Decline)		
		ACK	→	
Comments	Check:	Is the P-Asserted-Identity pres	ent?	
	Check: Is the communication rejected by sending a 603 (Decline) final			g a 603 (Decline) final
	response sent to user A?			
	Repeat t	his 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.		
Configuration	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 a="" of="" user=""> Privacy: id</uri>		
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 a="" of="" user=""> Privacy: id 433 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause = 21</uri>		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 603 Decline (REL)		
Comments	ACK → Check: Is the P-Asserted-Identity present?		
Comments	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 no	ot 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</th--></networkindicator></cug>		
	<cuginterlockbinarycode>0F03</cuginterlockbinarycode>		
	<cugcommunicationindicator>10</cugcommunicationindicator>		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network	(B)	
(10000000000000000000000000000000000000	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' eler	ment as	
	a 'cug' child element?		
	Check: Contains the XML body in the INVITE a 'cugInterlockBinaryCo	ode'	
	element as a 'cug' child element?	е	
	Check: Contains the XML body in the INVITE a 'cugCommunicationIn	idicator	
	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 cugInterlockBina	aryCode	
	element value are examples.	ii yooue	
	Giernent 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		ing user -OA to terminating user no CUG.		
rest purpose	Originat	ing user OA to terminating user no ooo.		
	An origin	ating 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	,		
Configuration	Originat	ing user: CUG, outgoing access not allowed		
SIP Parameter	INVITE:			
	Conte	ent-Type: application/vnd.etsi.cug+xml		
		ent-Disposition:;handling= required		
	<:0	<:cug>		
		networkIndicator>01		
		networkIndicator>23		
	<uginterlockbinarycode>0F03</uginterlockbinarycode>			
	<cugcommunicationindicator>11</cugcommunicationindicator>			
	<cl< th=""><th>Jg></th></cl<>	Jg>		
Message flow		lutana anno attan lutanfa a		
SIP (Network A)		Interconnection Interface SIP (Network B) INVITE →		
	←	403 (Forbidden)		
	•	ACK		
Comments	Check:	Is the Content-Type in The INVITE set to		
Comments	Officer.	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		
	Check:	Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network?		
	Check:	Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network? his test in reverse direction.		
	Check:	Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network?		

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		ing user -OA to terminating user -IA.	
	An origin	ating user in a CUG Outgoing Access not allowed calls to a user in the	
		JG Incoming Access not allowed. The session establishment is	
	successf	ul.	
Configuration		ing user: CUG, outgoing access not allowed	
	Terminating user: CUG incoming access not allowed		
	User in n	etwork A and user in network B are in the same CUG	
SIP Parameter	INVITE:		
		ent-Type: application/vnd.etsi.cug+xml	
	Conte	ent-Disposition:;handling= required	
	<cl< th=""><th></th></cl<>		
		networkIndicator>01	
	<networkindicator>23</networkindicator>		
	<cuginterlockbinarycode>0F03</cuginterlockbinarycode>		
	<cugcommunicationindicator>11</cugcommunicationindicator>		
Manager days	<cl< th=""><th>lg></th></cl<>	lg>	
Message flow SIP (Network A)		Interconnection Interface SIP (Network B)	
SIF (Network A)		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?	
		nis test in reverse direction.	
	NOTE:	The networkIndicator element value and the cugInterlockBinaryCode	
	1	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	Originat	ing user in a CUG to terminating user -IA.	
		_	
		ating user in a CUG calls to a user in a differ	
		ed. The session establishment is not succes	sful, a 403 (Forbidden)
	response		0110
Configuration		etwork A and user in network B are not in the	
0.0		ting user: CUG incoming access not allowed	a
SIP Parameter	INVITE:		
		ent-Type: application/vnd.etsi.cug+xml	
	Conte	ent-Disposition:;handling= requiredv	
		10>	
	<cug></cug>		
	<networkindicator>01 <networkindicator>23 <cuginterlockbinarycode>0F03</cuginterlockbinarycode> <cugcommunicationindicator> <cug></cug></cugcommunicationindicator></networkindicator></networkindicator>		
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 'cu	
	Check:	Contains the XML body in the INVITE a 'ne	tworkIndicator' element as
		a 'cug' child element?	
	Check:	Contains the XML body in the INVITE a 'cu	gInterlockBinaryCode'
		element as a 'cug' child element?	
	Check:	Contains the XML body in the INVITE a 'cu	
		element set to '10' or '11'as a 'cug' child ele	
	Check:	Is the session setup rejected? A 403 (Forbi	dden) final response is sent
		by the terminating network?	
		his test in reverse direction.	a accordant and a de Disa and Co. I
	NOTE:	The networkIndicator element value and the	ne cuginteriockbinaryCode
	1	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 cua	007	
Test case group	SS_cug_007 SIP-SIP/Service/CUG		
Reference	4.5.2.4/[13]		
SELECTION EXPRESSION	SE 34	0]	
Test purpose		ing user -OA, network B does not support CUG.	
l est purpose	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		
	Conte	ent-Disposition:;handling= required	
	 <cug></cug>		
	1	networkIndicator>01	
	<pre><networkindicator>23</networkindicator>0F03 <cugcommunicationindicator>10</cugcommunicationindicator> <cug></cug></pre>		
Message flow	\ 00		
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'	
	01	element set to '11' as a 'cug' child element?	
	Check:	Is the session setup rejected by sending an unsuccessful final	
	Popost #	response?	
	NOTE:	nis test in reverse direction. The networkIndicator element value and the cugInterlockBinaryCode	
	NOIL.	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			
Configuration	in the encapsulated IAM sent to Network B.			
Configuration SIP Parameter	User in PSTN/PLMN part of Network A in a CUG outgoing access allowed No. (U.E.)			
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			
	333			
Message flow	[any boundary name]			
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			
Message flow SIP (Network A)	[any boundary name] 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.			

Test case number	SS_cug_010			
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 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.			
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 			
SIP Parameter	User A and User B are in the same CUG INVITE			
Message flow	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]			
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 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.			

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 		
SIP Parameter	User A and User B are in the same CUG 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 →		
Comments	← 180 Ringing User A in the PSTN/PLMN part of Network A calls user B in Network B		
Comments	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		
	 User in PSTN/PLMN part of Network B in a CUG incoming access not allowed 		
SIP Parameter	User A and User B are in different CUG INVITE		
SIF Farameter	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 →		
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	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 allow	wed.		
	User A is located in Network A. User B is located in the PSTN/PLMN p SIP-I - ISUP/BICC interworking applied. Ensure that when user A calls 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 INVITE →	rk B)		
	← 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></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>'?</urn:alert:service:call-waiting>		
	Repeat this test in reverse direction.		

	100	200		
Test case number	SS_cw_002			
Test case group	SIP-SIP/Service/CW			
Reference	4.5.5.2/[1	l 5]		
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	uriavaliai	ne) response toward user A and the	ne Neason	Header Held is set to 19.
SIP Parameter	180: Alert-Info: <urn:alert:service:call-waiting></urn:alert:service:call-waiting>			
	480: Reason: Q.850 ;cause=19			
Message flow				
SIP (Network A)	_	Interconnection Interface INVITE	→	SIP (Network B)
	←	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>'?</urn:alert:service:call-waiting>			
	Check:	Sheck: 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.			

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 →			
	4 180 Ringing			
0	Apply post test routine			
Comments	Check: Is an ISUP/BICC ACM present in the 180 provisional response and			
	the Congris notification is not to half is a waiting call?			
	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.		
Tool parpood	Touppoin our Huming muloudon in 100 min choupsulated of O.		
	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		
Message flow	call is a waiting call		
SIP (Network A)	Interconnection Interface SIP (Network B)		
Oii (Network A)	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 LIBI in the Refer To header in set to 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	
		ddress 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 a	ddress of user C.	
Configuration			
SIP Parameter	REFER: Request URI address of user I		
	Refer-To: <uri ect-as="" of="">; metho</uri>	d=invite	
	INVESTA De successibility de deserva es FOT	A C	
	INVITE1 Request URI address of ECT-	AS	
	INVITE2: Request URI address of user	С	
Message flow	INVITEZ. Request Orti address of dser		
SIP (Network A)	Interconnection Interface	SIP (Network B)	
,	rmed session is established between u	` ,	
	rmed session is established between ι		
Us	er A invokes ECT to tr <mark>ansfer</mark> the sessi		
	REFER	→	
	202 Accepted		
	← NOTIFY (100)	→	
	200 OK NOTIFY	7	
CA	SE Blind transfer		
0,1	BYE (A-B)	→	
	€ 200 OK BYE	-	
	<mark>INVITE</mark> 2 (user C)	→	
	€ 200 OK INVITE	•	
	ACK	7	
	200 OK INVITE	→	
	← ACK ← NOTIFY (200)		
	200 OK NOTIFY	→	
	200 01(1001111	-	
CAS	E 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 tl	nis test in reverse direction.

Test case number	SS oot (003
	SS_ect_0	
Test case group		Service/ECT
Reference	4.5.2/[11]	
SELECTION EXPRESSION		A] SE37 AND [Network A] SE 11 AND [Network A] SE 50
Test purpose		ative transfer using the REFER method. s located in network A, user B and user C are located in network B. User
		s 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
	į	is present containing the session identifiers of the session A - C.
Configuration		
SIP Parameter		Request URI address of user B
	Refer	r-To: <uri ect-as="" of="">; method=invite</uri>
	IND/ITE	De succest LIDI e delecce et FOT AO
	INVITE	Request URI address of ECT-AS
	INI\/ITE2·	: Request URI address of user C
		ire: replaces
		aces: <session a-c=""></session>
Message flow	rtopic	2000. 10000101171 02
SIP (Network A)		Interconnection Interface SIP (Network B)
	med sess	sion is established between user A and user B
		sion is established between user A and user C
		kes ECT to transfer the session to user C
		REFER →
	←	202 Accepted
	←	NOTIFY (100)
		200 OK NOTIFY →
	_	
	←	INVITE 1 (ECT-AS)
	,	INVITE2 (user C) →
	←	200 OK INVITE
		ACK ->
	←	200 OK INVITE →
	-	ACK NOTIFY (200)
	~	200 OK NOTIFY →
		200 01(1011) 1
		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 (and a Penlaces header is present contains the
		address of user C and a Replaces header is present contains the
	Oh a -l-	session identifiers of the session A-C?
	Check:	session identifiers of the session A-C? Is the session A - B and the session A - C terminated?
	Check:	session identifiers of the session A-C?

Test case number	SS_ect_0	003	
Test case group		Service/ECT	
Reference	4.5.2/[11]	, 4.7.2.9.7/[20]	
SELECTION EXPRESSION		A] SE37 AND NOT [Network A] S	SE 12 AND [Network A] SE 49
Test purpose		sured transfer using the 3pcc m	
			ser C are located in network B User A
		ECT to transfer a session with use	
	_	Ensure that the network A establis	
		Ensure that the network A sends a	•
	b	etween user A and user B (SDP:	IP address, port and codec).
Configuration			
SIP Parameter	INVITE1	Request URI address of user C	
	INIVITEO.	Descript LIDI address of year D	
	SDP	Request URI address of user B	
		:IN IP4/6 [new IP address]	
		=audio [new port] RTP/AVP [new	codec list1
		-audio [new port] KTT /AVT [new	codec listj
Message flow	μ α-	-[new duribates]	
SIP (Network A)		Interconnection Interface	SIP (Network B)
	rmed sessi	ion is established between user	
Us	er A invok	es 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	7
Comments	Check:		network A to user C to establish a
Comments	OHECK.	dialogue between network A and	
	Check:		rk A to user B update the session
	JIICOK.	parameter in the SDP?	Tit / to door b apadio the sossion
	Repeat th	nis test in reverse direction.	
	i. topout u	teet reverse an eetieri	

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.
rest purpose	Consultative transfer using the spec 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 hetween year A and year C (SDR IP address, part and sedes)
Configuration	between user A and user C (SDP: IP address, port and codec).
Configuration	INVITED Degrees LIDI address of year C
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	a-[new authbutes]
SIP (Network A)	Interconnection Interface SIP (Network B)
,	med session is established between user A and user B
	med session is established between user A and user C
	er 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.
L	

Test case number	SS_ect_0	005
Test case group		- /Service/ECT
Reference	5.4.3.2/[2	24]
SELECTION EXPRESSION		(A] SE 17 AND SE 47 AND SE 60
Test purpose		pport. Call Transfer invoked in active state, call was previous on
Tool parpood	HOLD.	pport. Gail Transfer invented in delive state, Gail was provided on
		UP - SIP-I interworking applies in the originating network User A and C ted in network A and user B is located in network B.
	Ensure th	that an User A can successfully invoke the ECT supplementary service
	and trans	sfer the call with User B to User C in active state.
Configuration	User A is	s subscribed to the Explicit Call Transfer supplementary service
SIP Parameter	INVITE	· · · · · · · · · · · · · · · · · · ·
	Co	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
	F.A	AC
		Generic Notification
		Call transfer active
		Call transfer number
	[a	-[any boundary name]
Message flow		
SIP (Network A)		Interconnection Interface SIP (Network B)
		stablished between user A and user B and set on hold
U:	ser A invok	kes 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		on from User A to User B is already established
		sets the User B on hold
		nvokes the ECT service
	Check:	\ I
		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 th	this test in reverse direction.

Test case number	SS_ect_0	006	
Test case group		Service/ECT	
Reference	5.4.3.2/[2	24]	
SELECTION EXPRESSION		A] SE 17 AND SE 47 AND SE	60
Test purpose			alerting state, call was previous on
	HOLD.		3 ,
	BICC/ISI	JP - SIP-I interworking applies in	n the originating network User A and C
	are locate	ed in network A and user B is lo	cated in network B.
			nvoke the ECT supplementary service
	and trans	sfer 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	·	
	C	ontent-Type: multipart/mixed;bo	undary=[any boundary name]
	!	[any boundary name]	
		ontent-Type: application/sdp	
	a:	=sendrecv	
	!	[any boundary name]	
		ontent-Type: application/isup;ve	ersion=itu-t92
		ontent-Disposition: signal;handl	
	F/	AC	•
		Generic Notification	
		Call transfer alerting	
		Call transfer number	
	!	[any boundary name]	
Message flow		<u> </u>	
SIP (Network A)		Interconnection Interface	SIP (Network B)
A confirmed se	ssion is es	stablished between user A and	d user B and set on hold
Us	er A invok	es ECT to transfer the session	n 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		n from User A to User B is alrea	dy established
		ets the User B on hold	
		n from User A to User C is alrea	dy established
		vokes the ECT service	
	Check:	Is (optional) an INFO request	is sent from Network A to Network B
			present the Loop prevention indicator
		set to 'request'?	
	Check:		is sent from Network A to Network B
			present the Loop prevention indicator
		set to 'response'?	
	Check:		st sent and an ISUP FAC message is
			notification indicator is set to 'Call
			n the media stream is set to 'sendrecv'?
	Check:		sent and an ISUP FAC message is
			notification indicator is set to 'Call
			s an INVITE request sent and the media
		stream is set to 'sendrecv' to re	
	NOTE:		INVITE request is Equal to the content
		of the FAC in the INFO reques	·
	Repeat th	his test in reverse direction.	
<u> </u>	<u></u>		

Test case number	SS ect 007		
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.		
l est pui pose	on -1 support. Can 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.		
Configuration	User A is subscribed to the Explicit Call Transfer supplementary service		
SIP Parameter	INFO		
Sir raiametei	Content-Type: application/isup;version=itu-t92		
	Content-Type: application/isup,version=itu-ts2 Content-Disposition: signal;handling=required		
	FAC		
	Generic Notification		
	Call transfer active		
	Call transfer number		
Message flow	Can transfer transfer		
SIP (Network A)	Interconnection Interface SIP (Network B)		
	med session is established between user A and user C		
Use	er 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		
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 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.		

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.		
	DIGO (IGUID GID III A LID		
	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)		
	the early dialogue is established between user A and user C		
	er 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 INFÓ		
	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.		
	Inopodi uno tost 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.		
Configuration	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. User B is subscribed to the MCID service		
SIP Parameter	INFO: <:mcid>		
	<pre><:request> <:McidRequestIndicator>01<!--:McidRequestIndicator--> <:HoldingIndicator ><!--:HoldingIndicator--> <!--:request--> <!--:mcid--></pre>		
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.
Tool purpose	Notwork B condo a mois request, mois 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 →
	<mark>INFO</mark> →
	← 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 OrigPartyldentity 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.
	INCOMPT THE TOET IN TOVOTED DIFFCTION

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 INVITE INFO(IDR) 200 OK INFO Timeout T₀-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.
l'est pui pose	on -1 support. Network b serius a moib request, moib 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
Comiguration	User A is a ISDN or POTS user in the PSTN of network A
SIP Parameter	INFO:
on randicier	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	Content-Disposition: Signal, nandling-required
	IDR
	MCID request indicators
	MCID request indicator
	MCID requested
	WOID requested
	INFO:
	Content-Type: application/isup;version=itu-t92
	Content-Type: application/isup,version=itu-is2
	Content-Disposition. Signal, nandling-required
	IRS
	MCID response indicators
	MCID response indicator
	MCID included
	Calling party number
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
on (noment)	INVITE -
	← INFO(IDR)
	200 OK INFO →
	INFO(IRS)
	€ 200 OK INFO
	€ 180 Ringing
	Apply post test routine
Comments	
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'?
	Repeat this test in reverse direction.

7.1.5.13 Message Waiting Indication (MWI)

Test case number	SS_r	mwi_001		
Test case group	SIP-S	SIP-SIP/Service/MWI		
Reference	4.7.2	4.7.2/[16]		
SELECTION EXPRESSION	Netv [Netv	[Network A] SE 39 AND [Network B] SE 39		
Test purpose	Initia	Initial subscription of a Voicemail box.		
		The Voicemail owner is in network A, his Voicemail box is located in network B.		
		re that a Voicemail owner is able to a	activate his	Voicemail box.
Configuration		Voicemail in network B		
		email owner in network A		
SIP Parameter	SUB	CRIBE		
		Event: message-summary		
		Expires: [any value]		
		Accept: application/simple-mes	ssage-sumn	nary
	NOT	IFV		
	NOT		roo lonuud	luo1
		Subscription-State: active;expire Event: message-summary	res=tany va	luej
Message flow		Event. message-summary		
SIP (Network A)		Interconnection Interface		SIP (Network B)
(SUBCRIBE	→	J. (1311131112)
	(200 OK SUBSCRIBE		
	←	NOTIFY		
		200 OK NOTIFY	→	
	←	200 OK BYE		
	_			
	←	NOTIFY	_	
		200 OK NOTIFY	→	
	01	Apply post test routine		7 ()/ : 71 :
Comments	Check:	Is it possible for a user in netwo	ork A to sub	scribe to a Voicemail box in
	Check:	network B? Is the Event header in the SUB	CDIRE cot t	o 'massaga-summary'?
	Check:	Is the Accept header in the SUI		
	OHECK.	message-summary'?	DOMIDE SEL	to application/simple*
	Check:	Is the Event header in the NOT	TFY is set to	'message-summary'?
		his test in reverse direction.		moodago odminary .
ļ	. topoat t			

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	
	Check: Is the Event header in the NOTIFY set to 'message-summary'? 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'? Check: Contains the MIME body the optional header 'Voice-Message'? Check: Contains the MIME body the optional header 'Voice-Message'? Check: Contains the MIME body the optional header 'Voice-Message'?	

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

Test case number	SS cc 0	01		
Test case group		Service/CC		
Reference	4.5.4.3/[1			
SELECTION EXPRESSION	-	•		
Test purpose	[Network A] SE 40 AND [Network B] SE 40 Indicating of CCBS possible.			
rest purpose	muicatii	ig of CCB3 possible.		
	Lloor A in	located in network A and user B is	a located in naturals D	
		when user A calls user B and user I	• · · · · · · · · · · · · · · · · · · ·	
	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</sip:ue-b>			
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:			
		network B.		
	Check:			
		'call-completion' and the m parameter set to 'BS'.		
	Repeat th	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</sip:ue-b>		
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
Took marked as	[Network B] SE 41) Invocation of CCBS or CCNR.
Test purpose	Invocation of CCBS of 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</a-as></ue-b></ue-a>
	NOTIFY sip:A-AS Event:call-completion Content-Type: application/call-completion state: queued service-retention
Message flow	
SIP (Network A) An indicatio	Interconnection Interface SIP (Network B) n 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.
	1.10 COLVED TOTAL THE COLVEN TO BOOK OF THE COLVEN OF THE

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 480 Temporarily Unavailable final response is sent.	
Configuration		
SIP Parameter	SUBSRIBE sip:B-AS;m=BS or m=NR	
	From: <ue-a></ue-a>	
	To: <ue-b></ue-b>	
	Contact: <a-as></a-as>	
	Event:call-completion	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)
An indicatio	n whether CCBS or CCNR is possible is sent by network B SUBSCRIBE →	
	-	
Comments	2 100 (Tomporaniy Ghavanasio)	
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 C	CBS
	request or set to 'NR' in case of CCNR?	CDS
	Check: Is a 480 Temporarily Unavailable sent from network B indicates	the
	CCBS or CCNR request is unsuccessful e.g. CC queue is full?	1116
	Repeat this test in reverse direction.	
	Tropodi uno toti in fovoloti dilottori.	

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) A	Interconnection Interface 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 posting	
Comments	Apply post test routine Check: Is a NOTIFY request is sent to network A and the Event header is set	
Comments	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
OLLEGION EXI NEGGION	[Network B] SE 41)
Test purpose	User A is unavailable while CC recall is performed.
	Con 71.0 ana 1ana 10.0 con 10.
	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
0	procedure.
Configuration	NOTIFY size O. A.C.
SIP Parameter	NOTIFY sip:O-AS
	Event:call-completion Content-Type: application/call-completion
	state: ready
	State. Today
	PUBLISH sip B-AS
	To: SIP 2
	Event: presence
	Content-Type: application/pidf+xml
	xml version="1.0" encoding="UTF-8"?
	<pre><pre><pre><pre><pre><pre><pre><pre></pre></pre></pre></pre></pre></pre></pre></pre>
	<status></status>
	<basic>closed</basic>
	DUDUCU ain D. A.C.
	PUBLISH sip B-AS To: SIP 2
	Event: presence
	Content-Type: application/pidf+xml
	xml version="1.0" encoding="UTF-8"?
	<pre><pre><pre><pre><pre><pre></pre></pre></pre></pre></pre></pre>
	- <status></status>
	<basic>open</basic>
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
	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
Comments	

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	OID (Not on 1 D)
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		
Configuration	encapsulated IAM of the initial INVTE request. User A is subscribed to the User-to-User service 1 implicit request		
SIP Parameter			
SIP Parameter	INVITE:		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	IAM		
	User-to-user Information		
	User Information		
Message flow	User information		
SIP (Network A)	Interconnection Interface SIP (Network B)		
on (noment)	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
	User information
	180
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ACM
	User-to-user Information
	User Information
Message flow	Interconnection Interfers CID (Naturals D)
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?
Commones	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) →
0	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.
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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 routing
Comments	Apply post test routine Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request?
Comments	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',
	prepeat this test in reverse unection.

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)
Comments	Apply post test routine 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:		
on randineter	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	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)		
	← 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.		
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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	User B is not subscribed to the User-to-User service 2 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 purpose SiP-I support: Indicating of User-to-User service 2 essential rejection. BICC/ISUP - SIP-I interworking applies in the originating and terminating netwo 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 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 INVITE (IAM) ← 500 Server Internal Error (REL) ACK Aply post test routine Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request? Check: Is an 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?	Test case number	SS_uus_010	
Test purpose ((Network A) SE 17 AND SE 47) AND ((Network B) SE 17 AND SE 47) AND SE 63 SIP-I support: Indicating of User-to-User service 2 essential rejection. BICC/ISUP - SIP-I interworking applies in the originating and terminating netwo 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 encapsulated ISUP/BICC REL is present, the Cause value is set to #29 or #69. Configuration	Test case group	SIP-SIP/SIP-I/UUS	
Test purpose ((Network A) SE 17 AND SE 47) AND ((Network B) SE 17 AND SE 47) AND SE 63	Reference	6.11.2, 7.1/[24]	
BICC/ISUP - SIP-I interworking applies in the originating and terminating netwo 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 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 INVITE (IAM) Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value #29 or #69 Message flow SIP (Network B) Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value #29 or #69 Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t	SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63	
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 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 INVITE (IAM) Cause value #29 or #69 Message flow SIP (Network B) Interconnection Interface INVITE (IAM) Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request? Check: Is an ISUP/BICC IAM encapsulated in the senapsulated ISUP/BICC IAM set to 'Request', 'service 1', 'essential'? Check: Is an ISUP/BICC IAM et no repeated in the 500 response?	Test purpose	SIP-I support: Indicating of User-to-User service 2 essential rejection.	
INVITE: Content-Type: application/isup;version=itu-t92		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	
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 INVITE (IAM) SIP (Network A) Interconnection Interface INVITE (IAM) 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?			
Content-Disposition: signal;handling=required REL Cause value #29 or #69 Message flow SIP (Network A) Interconnection Interface INVITE (IAM) SIP (Network B) INVITE (IAM) 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?	SIP Parameter	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM User-to-user Indicator Request service 2 essential	
#29 or #69 Message flow SIP (Network A) Interconnection Interface INVITE (IAM) SOU 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?		Content-Disposition: signal;handling=required	
SIP (Network A) Interconnection Interface INVITE (IAM) Solution 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?			
INVITE (IAM)			
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?	SIP (Network A)	· · · · · · · · · · · · · · · · · · ·	
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 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?		Apply post test routine	
Repeat this test in reverse direction.	Comments	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?	

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	
0	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.	

Test case number	SS_uus_012		
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 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.		
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	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		
Message flow	User Information		
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE (IAM) → 180 Ringing (ACM) 200 OK INVITE (ANM) ACK →		
	INFO (USR) → 200 OK INFO INFO (USR) 200 OK INFO → 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'? 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'? Charles on ISLID/RICC LISP encappealeted in the INFO message cent from		
	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_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		
GEEEGHON EXI REGGION	63		
Test purpose	SIP-I support: Indicating of User-to-User service 3 not essential rejected in		
rest purpose	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	Lacon and the form		
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE (IAM)		
	← 180 Ringing (ACM) ← 200 OK INVITE (ANM)		
	200 011 11111 (7 11 1111)		
	ACK → Apply post test routine		
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request?		
Comments	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.		
L	propout and tool in tovorso 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.	
	DIOCHOLID OID LA	
	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 →	
Comments	Apply post test routine Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request?	
Comments	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-Type: application/isup,version=fitu-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	Interconnection Interfere		
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		
	Apply post test routille		

Comments	A sessio	n 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 t	his test in reverse direction.

-	100		
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
0	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-Type: application/isup,version=itu-tez
	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.		
	DIOC/IOLID		
	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		
Managara flavo	[any boundary name]		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)		
SIF (Network A)	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-Type: application/isup,version=itu-ts2 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.			

Test case number	SS_tp_002
Test case group	SIP-SIP/SIP-I/TP
Reference	5.4.3.2, 6.11.2, 6.11.2/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 64
Test purpose	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.
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
	BYE Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Location public network serving remote user Cause value 102
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
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 REL message encapsulated in the BYE request and the Cause value set to #102?
	Repeat this test in reverse direction.

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.		
	the INVIT	ttempts to call user B ported to network B. Ensure that the userinfo in FE contains a destination number in the global number format, an 'rn' er containing the Number Portability Routing Number in a global number ith hex digits and optional the 'npdi' parameter.		
Configuration				
SIP Parameter	sip: +	Request line <cc> <ndc> <sn>[;npdi][; rn=(Number portability routing number)] ehostname>;user = phone SIP/2.0</sn></ndc></cc>		
Message flow SIP (Network A)		Interconnection Interface SIP (Network B)		
SIF (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 th	his 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</hostname></sn></ndc></cc>
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_00	1		
Test case group	SIF	P-SIP/AC	COUNTING		
Reference					
SELECTION EXPRESSION					
Test purpose	Со	mpariso	n of Charging Data Reco	ords > 1 s.	
	the	active s	ession stored in the CDR of	th a duration > 1 s. Verify the duration of of both networks compared with the ow at the Interconnection Interface.	
Configuration					
SIP Parameter					
Message flow					
SIP (Network A)		lı C	nterconnection Interface INVITE 180 Ringing	SIP (Network B) →	
		÷	200 OK INVITE		
		-	ACK	→	
			Communication		
			BYE	→	
		←	200 OK BYE		
Comments	1. 2. 3.	Verify is	a call from network A to ne s the session confirmed. ate the session after 5 s.	twork B.	
	4.	Determ	nine the duration of the ses	ssion from the trace of the call monitor.	
	5.	Compa	re the following information	n elements indicated in the CDR's of both	
		networl	ks:		
		• C:	alling party number		
			alled party number		
	 timestamp 				
			allduration		
	callsetuptime (optional)				
	6.	Check trace.	the duration indicated in th	ne CDR against the duration in the call	
	7.	Repeat	this test in reverse direction	on.	

Test case number	SS acc 002		
Test case group	SIP-SIP/ACCOUNTING		
Reference	OII -OII /ACCOUNTING		
SELECTION EXPRESSION	O (OL D. (D L d.		
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	Setup a call from network A to network B.		
	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	SII -OII /ACCOUNTING		
SELECTION EXPRESSION			
Test purpose	Comparison of Charging Data Records > 15 min.		
	Accounting of a confirmed according with a duration of > 15 min. Verify the		
	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	the duration in the monitored message now at the interconnection interface.		
Configuration			
SIP Parameter			
Message flow	leteres were the eleteric and the second of		
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE →		
	← 180 Ringing		
	← 200 OK INVITE		
	ACK →		
	Communication		
	BYE →		
	← 200 OK BYE		
Comments	Setup a call from network A to network B.		
	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		
	SIP-SIP/ACCOUNTING		
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)		
Comments	INVITE 180 Ringing 200 OK INVITE ACK Communication BYE 200 OK BYE 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.		

Test case number	SS_acc_005
Test case group	SIP-SIP/ACCOUNTING
Reference	SIF-SIF/ACCOUNTING
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	 Setup a call from network A to network B. Verify is the session confirmed. Terminate the session after 35 min. Determine the duration of the session from the trace of the call monitor. Compare the following information elements indicated in the CDR's of both networks: calling party number called party number timestamp callsetuptime (optional) Check the duration indicated in the CDR against the duration in the call trace. 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)		
Comments	 Setup a call from network A to network B. Verify is the session confirmed. Terminate the session at the earliest 61 min and at the latest 119 min. Determine the duration of the session from the trace of the call monitor. Compare the following information elements indicated in the CDR's of both networks: calling party number called party number timestamp callduration callsetuptime (optional) Check the duration indicated in the CDR against the duration in the call trace. Repeat this test in reverse direction. 		

Test case number	100 007		
	SS_acc_007		
Test case group	SIP-SIP/ACCOUNTING		
Reference			
SELECTION EXPRESSION			
Test purpose	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.		
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
, , ,	INVITE →		
	← 180 Ringing		
	← 200 OK IŇVIŤE		
	ACK →		
	Communication		
	BYE →		
	← 200 OK BYE		
Comments	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.		
	7. Repeat this test in reverse unection.		

Test case number	SS_acc_008
Test case group	SIP-SIP/ACCOUNTING
Reference	OII -OII /AGGGGITTING
SELECTION EXPRESSION	O COL COL D. C. D D D d
Test purpose	Comparison of Charging Data Records less than 1 s.
	A
	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
0	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	Setup a call from network A to network B.
	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.
	11. Nepeat this test in reverse unection.

Test case number	SS acc 009		
Test case group	SIP-SIP/ACCOUNTING		
Reference	Oil Oil Meddelltille		
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 INVITE		
Comments	 Setup a call from network A to network B. Verify is an early dialogue established. Terminate the early dialogue after 20 s. Determine the duration of the session from the trace of the call monitor. Compare the following information elements indicated in the CDR's of both networks: calling party number called party number timestamp callouration callsetuptime (optional) Check the duration indicated in the CDR against the duration in the call trace. Repeat this test in reverse direction. 		

7.4 Carrier Selection

Test case number	SS_csel_	001
Test case group	SIP-SIP/C	CS
Reference	5.7.1.10/[2	2]
SELECTION EXPRESSION	[Network	A] SE14 AND [Network B] SE15
Test purpose	User sele	ects an operator 'call-by-call'.
	User A an	nd user B are located in network A. Ensure that user A is able to call
	user B an	d user A is able to select network B as a selected carrier 'call-by-call'.
Configuration	User in ne	etwork A is not presubscribed
SIP Parameter		Request line
	sip: + <c0< th=""><th>C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc></th></c0<>	C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc>
		Request line
	sip	o: + <cc> <ndc> <sn>;npdi</sn></ndc></cc>
		[;rn= <number number="" portability="" routing="">]@<hostname>;</hostname></number>
		user=phone SIP/2.0
Message flow		Interconnection Interfere
SIP (Network A)		Interconnection Interface SIP (Network B) INVITE 1 →
	←	INVITE 1 7
	~	Apply post test routine
Comments	Check:	Is the 'cic' tel uri parameter present in the Request URI in the INVITE
Comments	Officer.	sent from network A to network B identifying the selected carrier?
	Check:	Is the 'npdi' parameter present in the Request URI of the INVITE
	0.100111	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:	
		regarding the end user charging in case of Carrier selection.
	Repeat th	is 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</hostname></sn></ndc></cc>
	INVITE: Request line
	sip: + <cc> <ndc> <sn>;npdi</sn></ndc></cc>
	[;rn= <number number="" portability="" routing="">]@<hostname>;</hostname></number>
	user=phone SIP/2.0
Maccago flow	
Message flow	Interconnection Interface SID (Network B)
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE 1 →
	INVITE 1 → INVITE 2
	INVITE 1 → INVITE 2 Apply post test routine
SIP (Network A)	INVITE 1 Apply post test routine Check: Is the 'cic' tel uri parameter present in the Request URI in the INVITE
SIP (Network A)	INVITE 1 Apply post test routine 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?
SIP (Network A)	INVITE 1 Apply post test routine 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?
SIP (Network A)	INVITE 1 Apply post test routine 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?
SIP (Network A)	INVITE 1 Apply post test routine 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?
SIP (Network A)	INVITE 1 Apply post test routine 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
SIP (Network A)	 INVITE 1 INVITE 2 Apply post test routine 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.
SIP (Network A)	 ← INVITE 1
SIP (Network A)	 INVITE 1 INVITE 2 Apply post test routine 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.

Test case number	SS_csel_	003
Test case group	SIP-SIP/C	S
Reference	5.7.1.10/[2	2]
SELECTION EXPRESSION	Network	A] SE14 AND [Network B] SE15
Test purpose	User is p	resubscribed to an operator unequal to B, and overrides the
	preselect	ion with call-by-call via operator B.
		d user B are located in network A. User A is preselected to a network
		network B. Ensure that user A is able to call user B and user A is able
		network B as a selected carrier 'call-by-call'. The preselected carrier is
	ignored.	
Configuration		etwork A is presubscribed to network B
SIP Parameter		equest line
	sip: + <c0< th=""><th>C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc></th></c0<>	C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc>
		lequest line
	sip	: + <cc> <ndc> <sn>;npdi</sn></ndc></cc>
		[;rn= <number number="" portability="" routing="">]@<hostname>;</hostname></number>
Manager	1	user=phone SIP/2.0
Message flow		Interconnection Interface SIP (Network B)
SIP (Network A)		Interconnection Interface SIP (Network B) NVITE 1 →
	←	INVITE 1 3
	•	Apply post test routine
Comments	Check:	Is the 'cic' tel uri parameter present in the Request URI in the INVITE
Comments		sent from network A to network B identifying the selected carrier?
		Is the 'npdi' parameter present in the Request URI of the INVITE
	Oncok.	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 th	is test in reverse direction.
	•	

Test case number	SS_csel_0	004
Test case group	SIP-SIP/C	S
Reference	5.7.1.10/[2	2]
SELECTION EXPRESSION	[Network]	A] SE14 AND [Network B] SE15
Test purpose	User is p	resubscribed to an operator not operator B, and overrides the
	preselect	ion with call-by-call via operator B.
		d user B are located in network A. User A is preselected to a network
		network B. Ensure that user A is able to call user B and user A is able
		network B as a selected carrier 'call-by-call'. The preselected carrier is
	ignored.	
Configuration		etwork A is presubscribed not to network B
SIP Parameter		equest line
	sip: + <c0< th=""><th>C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc></th></c0<>	C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc>
		equest line
	sip	: + <cc> <ndc> <sn>;npdi</sn></ndc></cc>
		[;rn= <number number="" portability="" routing="">]@<hostname>;</hostname></number>
Manager		user=phone SIP/2.0
Message flow		Interconnection Interface SIP (Network B)
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
Comments		sent from network A to network B identifying the selected carrier?
		Is the 'npdi' parameter present in the Request URI of the INVITE
		request sent from network B to network A?
		Is optional the 'rn' parameter present in the Request URI of the INVITE
		request sent from network B to network A?
		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 th	is 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 p	ating user in a CUG Outgoing Access not allowed preselected to B and calls to a user in the same CUG. The session establishment is	
Configuration		network A is presubscribed to network B network A are in the same CUG	
SIP Parameter	INVITE: Request line sip: + <cc> <ndc> <sn>@tariff.<hostname> user=phone SIP/2.0</hostname></sn></ndc></cc>		
		ent-Type: application/vnd.etsi.cug+xml ent-Disposition:;handling= required	
	<:C		
	<:C	:: cugCommunicationIndicator>11 : cugCommunicationIndicator ug>	
		Request line p: + <cc> <ndc> <sn@<hostname>;user=phone SIP/2.0</sn@<hostname></ndc></cc>	
		ent-Type: application/vnd.etsi.cug+xml ent-Disposition:;handling= required	
	c	ug>	
	<:C	cugCommunicationIndicator>11 : cugCommunicationIndicator	
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? Contains the XML body in the INVITE a 'cugCommunicationIndicator'	
	Check:	element set to '11' as a 'cug' child element? 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

Test case number	SS_sipc_001		
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 one single tariff.		
	3		
	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.		
Configuration			
SIP Parameter	INVITE:		
	Supported: 100rel		
	18x or 200 OK		
	Require: 100rel		
	ContentType: application/vnd.etsi.sci+xml		
	Content-Disposition: render; handling=optional		
	_		
	messageType		
	chargingControlIndicators		
	chargingTariff		
	tariffCurrency		
	currentTariffCurrency		
	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(crat)		
CASE B	200 OK INVITE(crgt)Apply post test routine		
Comments	Check: Is the supported header in the initial INVITE set to '100rel'		
Comments	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?		
İ	Repeat this test in reverse direction.		

Test case number	SS_sipc	002
Test case group		SIP_charging
Reference	B.2.3/[19	
SELECTION EXPRESSION	SE 16	1
Test purpose		Iful session from user A to user B via network B several tariffs in
rest purpose	one seq	uence.
	in case c	s located in network A and network B is responsible for charging (CDP) of carrier selection or service. Ensure that the network B sends a tariff
		on with several tariffs in a sequence covered in a XML MIME body in a provisional or successful final response.
Configuration		
SIP Parameter	INVITE: Supp	orted: 100rel
	18x or 20	00 OK
	Requ	ire: 100rel
		entType: application/vnd.etsi.sci+xml ent-Disposition: render; handling=optional
	mess crgt	ageType
		nargingControlIndicators
	CI	nargingTariff tariffCurrency
		currentTariffCurrency
		communicationChargeSequenceCurrency
		currencyFactorScale
		currencyFactor
		currencyScale
		tariffDuration
		subTariffControl
		communicationChargeSequenceCurrency
		currencyFactorScale
		currencyFactor currencyScale
		tariffDuration
		subTariffControl
		tariffControlIndicators
	01	riginationIdentification
		urrency (optional)
Message flow SIP (Network A)		Interconnection Interface SIP (Network B)
CASE A	←	INVITE → 18x(crgt)
OAGE A	•	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
	Check:	INVITE final response? Is the tariffCurrency element set to 'currentTariffCurrency'?
		Are there more than one communicationCharge
	OHECK.	SequenceCurrency elements present in the currentTariffCurrency
	Ola III	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'?
		Is the optional element ' currency ' set to 'EUR' if present?
		his test in reverse direction.
	ntopout	THE COST III TO FOLOO GILLOUIDII.

Test case number	SS_sipc_003		
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 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.		
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		
	callAttemptChargeCurrency		
	currencyFactor		
	currencyScale originationIdentification		
	currency (optional)		
Message flow	Curroney (optional)		
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 '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.		
	Troposit and toot in tovolog undomoni		

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
root purpose	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
	tariffControllndicators
	callSetupChargeCurrency
	currencyFactor
	currencyScale
	originationIdentification currency (optional)
Message flow	Currency (optional)
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_charging
Reference	B.2.3/[19	
SELECTION EXPRESSION	SE 16	
Test purpose		ful session from user A to user B via network B with a next tariff.
	in case of information	located in network A and network B is responsible for charging (CDP) f carrier selection or service. Ensure that the network B sends a tariff on with a next tariff and tariff switch over time covered in a XML MIME reliable provisional or successful final response.
Configuration	, , , ,	
SIP Parameter	INVITE: Suppo	orted: 100rel
	Conte	0 OK re: 100rel ntType: application/vnd.etsi.sci+xml nt-Disposition: render; handling=optional
	<mark>crgt</mark> ch	ageType argingControlIndicators argingTariff
		tariffCurrency currentTariffCurrency
		communicationChargeSequenceCurrency
		currencyFactorScale
		currencyFactor
		currencyScale tariffDuration
		subTariffControl
		tariffControlIndicators
		tariffSwitchCurrency
		nextTariffCurrency
		communicationChargeSequenceCurrency
		currencyFactorScale currencyFactor
		currencyScale
		tariffDuration
		subTariffControl
		tariffControlIndicators
	ori	tariffSwitchOverTime iginationIdentification
		rrency (optional)
Message flow	00	Tionoy (optional)
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'? Is the Require header in the response containing the tariff information
	Check:	set to '100rel'? Is the messageType ' crgt ' present in a 1xx provisional or a 200 OK
	Check: Check:	INVITE final response? Is the tariffSwitchCurrency element set to 'nextTariffCurrency'? Represents the currencyFactorScale in the
	Check:	communicationChargeSequenceCurrency element the next tariff? Is the time to change the tariff indicated in the tariffSwitchOverTime
	Check:	element? Is the optional element 'currency' set to 'EUR' if present?
	Repeat th	nis 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
Test purpose	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
	tariffOuration
	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
	_
	message Type
	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.

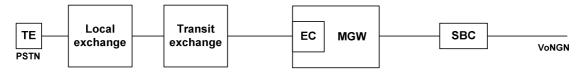


Figure 7.7-2: Reference configuration for PSTN/ISDN with classical access

7.7.1.3 NGN PSTN/ISDN access Configuration

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

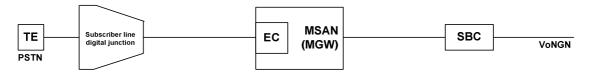


Figure 7.7-3: Reference configuration for NGN with PSTN/ISDN access

7.7.1.4 Access DSL Configuration

Figure 7.7-4 shows the xDSL access configuration.

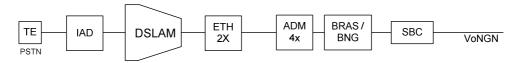


Figure 7.7-4: Reference configuration for DSL access

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 ($1\,400\,\text{km}/11\,\text{ms}$) are contained in clause 4 of TR 102 775 [i.3]

7.7.2 Test purposes for Quality of Service test

	<u></u>
Test case number	SS_qos_001
Test case group	SIP-SIP/QoS
Transmission Type:	Voice
Preconditions user	Reset Jitter Buffer 1 and Jitter Buffer 2 (e.g. by establishing a new call)
segment A:	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	Reset Jitter Buffer 1 and Jitter Buffer 2 (e.g. by establishing a new call)
segment B:	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 $D_{tr \ seg \ A-B} = (D_{sum \ seg \ A-B} - D_{JB1seg \ B} - D_{JB2segA})/2$ Loop in user segment A $D_{tr \ seg \ B-A} = (D_{sum \ seg \ B-A} - D_{JB1seg \ B} - D_{JB2segA})/2$
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	Reset Jitter Buffer 1 and Jitter Buffer 2 (e.g. by establishing a new call)
segment A:	Apply signal "single-talk" to Interface A and determine Delay D _{JB1} and D _{JB2}
Preconditions user	Reset Jitter Buffer 1 and Jitter Buffer 2 (e.g. by establishing a new call)
segment B:	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. Dtr-seg A-B = Dsum-seg A-B - DJB1seg B Dtr-seg B-A = Dsum-seg B-A - DJB2segA
Calling station	The amplitude of the tone is -16 dBm0
Called station	The amplitude of the tone is -16 dBm0

Annex A (informative): Bibliography

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History

Document history			
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