ETSI TS 101 585 V1.2.1 (2014-04)



Core Network and Interoperability Testing (INT); IMS interconnection tests at the lc Interface; Test Suite Structure and Test Purposes (TSS&TP)

Reference

RTS/INT-00096

Keywords

interconnection, SIP, testing, UNI

ETSI

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

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

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

Important notice

The present document can be downloaded from: http://www.etsi.org

The present document may be made available in electronic versions and/or in print. The content of any electronic and/or print versions of the present document shall not be modified without the prior written authorization of ETSI. In case of any existing or perceived difference in contents between such versions and/or in print, the only prevailing document is the print of the Portable Document Format (PDF) version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status.

Information on the current status of this and other ETSI documents is available at

http://portal.etsi.org/tb/status/status.asp

If you find errors in the present document, please send your comment to one of the following services: <u>http://portal.etsi.org/chaircor/ETSI_support.asp</u>

Copyright Notification

No part may be reproduced or utilized in any form or by any means, electronic or mechanical, including photocopying and microfilm except as authorized by written permission of ETSI.

The content of the PDF version shall not be modified without the written authorization of ETSI.

The copyright and the foregoing restriction extend to reproduction in all media.

© European Telecommunications Standards Institute 2014.
All rights reserved.

DECTTM, **PLUGTESTS**TM, **UMTS**TM and the ETSI logo are Trade Marks of ETSI registered for the benefit of its Members. **3GPP**TM and **LTE**TM are Trade Marks of ETSI registered for the benefit of its Members and of the 3GPP Organizational Partners.

GSM® and the GSM logo are Trade Marks registered and owned by the GSM Association.

Contents

Intelle	ectual Property Rights	5
Forew	vord	5
1	Scope	6
2	References	6
2.1	Normative references	
2.1	Informative references	
3	Definitions and abbreviations	
3.1	Definitions	
3.1.1	Conventions for representation of SIP/SDP information	
3.2	Abbreviations	
4	Test Suite Structure (TSS)	11
5	Declarations	11
5.1	Numbering Scheme	11
5.2	Reference configuration	
5.3	Selection of End Devices	12
6	Selection Expressions	13
7	Test purposes	14
7.1	Testing of SIP protocol requirements	
7.1.1	Test purposes for Basic call, Successful	
7.1.2	Codec negotiation	
7.1.3	Resource Reservation	
7.1.4	Test purposes for SIP-SIP, Basic call, Unsuccessful	50
7.1.5	Test purposes for Supplementary services	61
7.1.5.1	Test purposes for OIP	61
7.1.5.2	T T	
7.1.5.3	T T	
7.1.5.4	1 1	
7.1.5.5		
7.1.5.6		
7.1.5.6	• • • • • • • • • • • • • • • • • • • •	
7.1.5.6	··-	
7.1.5.6 7.1.5.6		
7.1.5.6 7.1.5.6		
7.1.5.0 7.1.5.7		
7.1.5. <i>7</i>		
7.1.5.9		
7.1.5.1		
7.1.5.1		
7.1.5.1	1 , ,	
7.1.5.1	Message Waiting Indication (MWI)	207
7.1.5.1	Completion of Communications to Busy Subscriber (CCBS), Completion of Communication	ons by
	No Reply (CCNR)	
7.1.6	Other PSTN services (SIP-I interworking)	
7.1.6.1		
7.1.6.2		
7.1.6.3		
7.2	Number Portability	
7.3 7.4	Accounting	
7.4 7.5	Carrier Selection Emergency call	
7.5 7.6	SIP Support of Charging	

7.7	Quality of Service	258
7.7.1	Reference Configurations	258
7.7.1.1	Backbone Configuration	
7.7.1.2	PSTN/ISDN classic access Configuration	258
7.7.1.3	NGN PSTN/ISDN access Configuration	
7.7.1.4	Access DSL Configuration	
7.7.1.5	Delay Values	
7.7.2	Test purposes for Quality of Service test	259
Annex A	(informative): Bibliography	261
History.		263

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: "Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards", which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (http://ipr.etsi.org).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

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

1 Scope

[10]

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

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.

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

2.1 Normative references

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

the following refer	enced documents are necessary for the application of the present document.
[1]	ETSI TS 129 165: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Inter-IMS Network to Network Interface (NNI) (3GPP TS 29.165 Release 10)".
[2]	ETSI TS 124 229: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229 Release 10)".
[3]	IETF RFC 4566 (2006): "SDP: Session Description Protocol".
[4]	IETF RFC 3261 (2002): "SIP: Session Initiation Protocol".
[5]	IETF RFC 3264 (2002): "An Offer/Answer Model with the Session Description Protocol (SDP)".
[6]	IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".
[7]	ETSI TS 124 607: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.607 Release 10)".
[8]	ETSI TS 124 608: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.608 Release 10)".
[9]	ETSI TS 124 604: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication Diversion (CDIV) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.604 version 10.3.0 Release 10)".

Network (CN) subsystem; Protocol specification (3GPP TS 24.605 Release 10)".

ETSI TS 124 605: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Conference (CONF) using IP Multimedia (IM) Core

- [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] Recommendation ITU-T V.152 (November 2004): "Procedures for supporting Voice-Band Data over IP Networks".
- [23] Recommendation ITU-T T.38 (September 2010, prepublished): "Procedures for real-time Group 3 facsimile communication over IP networks".
- [24] Recommendation ITU-T 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".
- [26] IETF RFC 4733: "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals".

[27] IETF RFC 4028: "Session Timers in the Session Initiation Protocol (SIP)".

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] Recommendation ITU-T 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:

BICC or ISUP specific terminology, references to Recommendation ITU-T Q.1902.2 [i.4]. For SIP and SDP specific terminology, references 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 ISUP exchange) 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 reacts 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 in capital.

EXAMPLE 1: INVITE, INFO.

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

EXAMPLE 2: To, From, Call-ID.

Where it is necessary to refer with finer granularity to components of a SIP message, the component concerned is identified by the ABNF rule name used to designate it in the defining RFC (generally 25/RFC 3261 [4]), in plain text without surrounding angle brackets.

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

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

EXAMPLE 4: sip: URI, tel: URI.

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

EXAMPLE 5: 100 Trying, 484 Address Incomplete.

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

EXAMPLE 6: 200 OK INVITE, 200 OK PRACK.

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

EXAMPLE 7: m=line, a=line.

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

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

3.2 Abbreviations

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

ACR Anonymous Communication Rejection

CB Communication Barring

CFB Communication Forwarding Busy

CCBS Completion of Communications to Busy Subscriber CCNR Completion of Communications by No Reply

CD Communication Deflection
CDIV Communication DIVersion
CDP Charging Determinating Point
CDR Communication Data Record

CFNL Communication Forwarding Not Logged in CFNR Communication Forwarding No Reply CFU Communication Forwarding Unconditional

CONF Conference
CUG Closed User Group
CW Communication Waiting

ECT Explicit Communication Transfer

FFS For Further Study

GW GateWay

HOLD Communication Hold

ISDN Integrated Services Digital Network

IUT Implementation Under Test

MCID Malicious Communication Identification

MWI Message Waiting Indication

NDUB Network Determined User Busy
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
UDUB User Determined User Busy

4 Test Suite Structure (TSS)

BCALL	successful	SS_bcall_xxx	
	Codec_Negotiation	SS_codec_xxx	
	Resource_Reservation	SS_resource_xxx	(
	unsuccessful	SS_unsucc_xxx	
SIP-SIP	Service	OIP	SS_oip_xxx
		OIR	SS_oir_xxx
		TIP	SS_tip_xxx
		TIR	SS_tir_xxx
		HOLD	SS_hold_xxx
		CFU	SS_cfu_xxx
		CFB	SS_cfb_xxx
		CFNR	SS_cfnr_xxx
		CFNL	SS_cfnl_xxx
		CD	SS_cd_xxx
		CONF	SS_conf_xxx
		ACR-CB	SS_acr-cb_xxx
		CUG	SS_cug_xxx
		CW	SS_cw_xxx
		ECT	SS_ect_xxx
		MCID	SS_mcid_xxx
		MWI	SS_mwi_xxx
		CC	SS_cc_xxx
	SIP-I	UUS	SS_uus_xxx
		SUB	SS_sub_xxx
		TP	SS_tp_xxx
		1	00_10_2000
	NubP	SS_NP_xxx	
	ACCOUNTING	SS_acc _xxx	
	CS	SS_csel_xxx	
	EmC	SS_ecall_xxx	
	SIP_charging	SS_sipc_xxx	
	SIP-SIP/QoS	SS_qos_xxx	

5 Declarations

5.1 Numbering Scheme

FFS.

5.2 Reference configuration

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

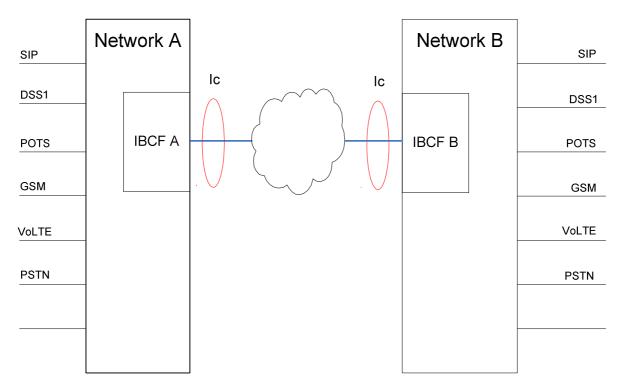


Figure 5.2-1: Reference configuration for the interconnection test

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 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 highly recommended due to the fact that the impact from an end device to another end device is important and will be marginal compensated by the network.

Which Test Purposes are possible to perform depends on the types of end devices used in the network. The table 5.3-1 gives an overview of end devices.

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: Overview of end devices

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.

• 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 for which role the Test purpose is applicable (Support Network A, Support Network B).

Before the start of the test, the table shall be filled out (yes/no) so the operators reply 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		
	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:	Void		
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?		
SE 16:	SIP Support of Charging is supported?		
SE 17:	The interworking ISUP - SIP I is performed in the network?		
	Supplementary services		
SE 18:	The network supports the Originating Identification Presentation (OIP)?		
SE 19:	The network supports the "Special arrangement" procedure for the originating user?		
SE 20:	The network supports the Originating Identification Restriction (OIR)?		
SE 21:	The Network supports the Terminating Identification Presentation (TIP)?		
SE 22:	The network supports the "Special arrangement" procedure for the terminating user?		
SE 23:	The Network supports the Terminating Identification Restriction (TIR)?		
SE 24:	The Network supports the session HOLD procedure?		
SE 25:	The network supports Communication Forwarding Unconditional (CFU)?		
SE 26:	The network supports Communication Forwarding Busy (CFB)?		
SE 27:	The network supports Communication Forwarding No Reply (CFNR)?		
SE 28:	The Network supports Communication Forwarding Not Logged in (CFNL)?		
SE 29:	The Network supports Communication Deflection?		
SE 30:	The Network supports the CDIV Notification procedure?		

	SELECTION EXPRESSION:	Support	Support
	 31: The Network supports conference (CONF)? 32: The Network supports the Communication Barring procedure (CB) (Black list for incoming calls)? 33: The Network supports the Anonymous Communication Rejection (ACR)? 34: The Network supports the Closed User Group (CUG)? 35: The Network supports the Communication Waiting (CW) service? 36: The Network supports the TAS-CW timer? 37: The Network supports Explicit Communication Transfer (ECT)? 38: The network supports Explicit Communication Identification (DI)? 39: The Network supports Message Waiting Indication (MWI)? 40: The Network supports Completion of Communications to Busy Subscriber (CCBS)? 41: The Network supports Completion of Communications by No Reply (CCNR)? 42: Void 43: The End device supports Fax transmission via G.711 codec? 44: The End device supports Fax transmission via W.152 codec? 45: The End device supports Fax transmission via m-line T.38 codec? 46: A SIP end device is used supporting an ISDN user equipment and the PSTN XML Schema is used? 47: End device is located in the PSTN or PLMN? 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? 49: The end device performs ECT using the 'Blind/assured transfer'? 50: The end device supported in the PSTN/PLMN part of the network? 51: The end device supported in the PSTN/PLMN part of the network? 52: CLIP/CLIR is supported in the PSTN/PLMN part of the network? 55: CDIV is supported in the PSTN/PLMN part of the network? 56: CONF/3PTY is supported in the PSTN/PLMN part of the network? 57: ACR is supported in the PSTN/PLMN part of the network? 58: CUB is supported in the PSTN/PLMN part of the network? 59: CW is supported in the PSTN/PLMN part of the network? 60: ECT is	Network A	Network B
SE 31:			
SE 32:			
SE 33:			
	The Network supports the Communication Waiting (CW) service?		
SE 38:	The network supports Malicious Communication Identification		
SE 31: The Network supports conference (CONF)? SE 32: The Network supports the Communication Barring procedure (CB) - (Black list for incoming calls)? SE 33: The Network supports the Anonymous Communication Rejection (ACR)? SE 34: The Network supports the Closed User Group (CUG)? SE 35: The Network supports the Communication Waiting (CW) service? SE 36: The Network supports the T _{AS-CW} timer? SE 37: The Network supports Explicit Communication Transfer (ECT)? SE 38: The network supports Explicit Communication Identification (MCID)? SE 39: The Network supports Message Waiting Indication (MWI)? SE 40: The Network supports Completion of Communications to Busy Subscriber (CCBS)? SE 41: The Network supports Completion of Communications by No Reply (CCNR)? Terminal capabilities SE 42: Void SE 43: The End device supports Fax transmission via G.711 codec? SE 44: The End device supports Fax transmission via W.152 codec? SE 44: The End device supports Fax transmission via W.152 codec? SE 45: The End device supports Fax transmission via W.152 codec? SE 46: A SIP end device is used supporting an ISDN user equipment and the PSTN XML Schema is used? SE 47: End device is located in the PSTN PLMN? SE 48: The terminating UE supports for PLMN? SE 49: 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 'Bilnd/assured transfer'? SE 50: The end device supported in the PSTN/PLMN part of the network? SE 51: The end device supported in the PSTN/PLMN part of the network? SE 52: CLIP/CLIR is supported in the PSTN/PLMN part of the network? SE 53: COLP/COR is supported in the PSTN/PLMN part of the network? SE 56: CONF/SPTY is supported in the PSTN/PLMN part of the network? SE 56: CONF/SPTY is supported in the PSTN/PLMN part of the network? SE 56: CUS is supported in the PSTN/PLMN part of the network? SE 56: CUS is supported in the PSTN/PLMN part of the network? SE 56: SUB i			
SE 40:			
SE 41:			
CE 40:	•		1
SE 40.			
SF 47·			
00.			
SE 49:			
SE 50:	The end device performs ECT using the 'Consultative transfer'?		
SE 51:			
	PSTN/PLMN Supplementary services		
	CLIP/CLIR is supported in the PSTN/PLMN part of the network?		
SE 32: The Network supports the Communication Barring procedure (CB) - (Black list for incoming calls)? SE 33: The Network supports the Anonymous Communication Rejection (ACR)? SE 34: The Network supports the Closed User Group (CUG)? SE 35: The Network supports the Communication Waiting (CW) service? SE 36: The Network supports Explicit Communication Transfer (ECT)? SE 37: The Network supports Explicit Communication Transfer (ECT)? SE 38: The network supports Message Waiting Indication (MWI)? SE 39: The Network supports Message Waiting Indication (MWI)? SE 40: The Network supports Completion of Communications to Busy Subscriber (CCBS)? SE 41: The Network supports Completion of Communications by No Reply (CCNR)? Terminal capabilities SE 42: Void 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 win-line T.38 codec? SE 46: A SIP end device supports Fax transmission via m-line T.38 codec? SE 46: The End device supports Fax transmission via m-line T.38 codec? 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 'Bind/assured transfer'? SE 50: The end device performs ECT using the 'Bind/assured transfer'? SE 51: The end device supported in the PSTN/PLMN part of the network? SE 55: CDIP/CLIR is supported in the PSTN/PLMN part of the network? SE 55: CDIP 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 59: CDIP is supported in the PSTN/PLMN part of the network? SE 59: CDIP is supported in the PSTN/PLMN part of the network? SE 59: CDIP is supported in the PSTN/PLMN part of the network? SE 59: CDIP is supported in the PSTN/PLMN part of the network? SE 59: CDIP is supported in the PSTN/PLMN part of th			
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 with 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.

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
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 user part 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 user part is in the format of a tel URI in global number format.
	Check: The host portion is set to the host name of the interconnected network.
	Check: The user parameter is set to phone.
	Repeat this test in reverse direction.
	Repeat this test with all chosen end devices.

Test case number	SS bcall 004
Test case group	BCALL/successful
Reference	5.10/[2]
Testspec Reference	
SELECTION EXPRESSION	SE 1
Test purpose	P-Charging-Vector header in the INVITE.
	Ensure that the P-Charging-Vector header is present in the INVITE establishes a communication between a user of network A and a user of network B and the 'icid-value' and the 'orig-ioi' parameter is present.
Configuration	
SIP Parameter	INVITE
	P-Charging-Vector: icid-value; orig-ioi
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE ->
	Apply post test routine
Comments	Establish a communication from network A to Network B
	Check: The P-Charging-Vector header contains the icid-value parameter.
	Check: The P-Charging-Vector header contains the orig-ioi parameter.
	Repeat this test in reverse direction.

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

Test case number	SS_bcall_006			
Test case group	BCALL/successful			
Reference	8/ [21]			
SELECTION EXPRESSION	[Network A] SE 3			
Test purpose	P-Early-Media header support indication in the initial INVITE request.			
	Ensure that the support of the P-Early. Media header is indicated in the initial INVITE request. A P-Early-Media header is present set to 'supported'.			
Configuration				
SIP Parameter	INVITE			
	P-Early-Media: supported SDP			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →			
	Apply post test routine			
Comments	Establish a communication from network A to Network B Check: Is a P-Early-Media header present in the INVITE request?			
	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	
Test purpose	P-Early-Media header supported in early dialogue.	
	Ensure that an early dialogue is established by sending a 183 Session Progress or 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	
	183	
	P-Early-Media: [any value authorizes early media]	
	SDP	
	OR	
	180	
	P-Early-Media: [any value authorizes early media] SDP	
Message flow	CDI	
SIP (Network A)	Interconnection Interface SIP (Network B)	
(,	INVITE >	
CASE A		
	← 183 Session Progress	
CASE B		
CASE B	← 180 Ringing	
CASE B		
CASE B Comments	Apply post test routine	
	Apply post test routine Establish a communication from network A to Network B Check: Is a 183 or 180 send from Network B to establish an early dialogue?	
	Apply post test routine Establish a communication from network A to Network B Check: Is a 183 or 180 send from Network B to establish an early dialogue? Check: Is an SDP present in the 183 as a SDP answer?	
	Apply post test routine Establish a communication from network A to Network B Check: Is a 183 or 180 send from Network B to establish an early dialogue? Check: Is an SDP present in the 183 as a SDP answer? Check: A bearer transmission is possible in backward directions.	
	Apply post test routine Establish a communication from network A to Network B Check: Is a 183 or 180 send from Network B to establish an early dialogue? Check: Is an SDP present in the 183 as a SDP answer? Check: A bearer transmission is possible in backward directions. NOTE: The absence of the direction parameter of an 'a' line represents the	
	Apply post test routine Establish a communication from network A to Network B Check: Is a 183 or 180 send from Network B to establish an early dialogue? Check: Is an SDP present in the 183 as a SDP answer? Check: A bearer transmission is possible in backward directions. NOTE: The absence of the direction parameter of an 'a' line represents the default value 'sendrecv'	
	Apply post test routine Establish a communication from network A to Network B Check: Is a 183 or 180 send from Network B to establish an early dialogue? Check: Is an SDP present in the 183 as a SDP answer? Check: A bearer transmission is possible in backward directions. NOTE: The absence of the direction parameter of an 'a' line represents the	
	Establish a communication from network A to Network B Check: Is a 183 or 180 send from Network B to establish an early dialogue? Check: Is an SDP present in the 183 as a SDP answer? Check: A bearer transmission is possible in backward directions. NOTE: The absence of the direction parameter of an 'a' line represents the default value 'sendrecv' NOTE: The presence of the P-Early-Media header in the INVITE request indicates the support of "early media Authorization" in the originating	
	Establish a communication from network A to Network B Check: Is a 183 or 180 send from Network B to establish an early dialogue? Check: Is an SDP present in the 183 as a SDP answer? Check: A bearer transmission is possible in backward directions. NOTE: The absence of the direction parameter of an 'a' line represents the default value 'sendrecv' NOTE: The presence of the P-Early-Media header in the INVITE request indicates the support of "early media Authorization" in the originating Network. NOTE: The presence of the P-Early-Media header in the 183 or 180 indicates the support of the P-Early-Media header and authorizes the media in the early dialogue	
	Establish a communication from network A to Network B Check: Is a 183 or 180 send from Network B to establish an early dialogue? Check: Is an SDP present in the 183 as a SDP answer? Check: A bearer transmission is possible in backward directions. NOTE: The absence of the direction parameter of an 'a' line represents the default value 'sendrecv' NOTE: The presence of the P-Early-Media header in the INVITE request indicates the support of "early media Authorization" in the originating Network. NOTE: The presence of the P-Early-Media header in the 183 or 180 indicates the support of the P-Early-Media header and authorizes the media in	
	Establish a communication from network A to Network B Check: Is a 183 or 180 send from Network B to establish an early dialogue? Check: Is an SDP present in the 183 as a SDP answer? Check: A bearer transmission is possible in backward directions. NOTE: The absence of the direction parameter of an 'a' line represents the default value 'sendrecv' NOTE: The presence of the P-Early-Media header in the INVITE request indicates the support of "early media Authorization" in the originating Network. NOTE: The presence of the P-Early-Media header in the 183 or 180 indicates the support of the P-Early-Media header and authorizes the media in the early dialogue	

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	
	404	
	181 D. Forly Modic: Jany valu authorized early modic!	
Message flow	P-Early-Media: [any valu authorizes early media]	
SIP (Network A)	Interconnection Interface SIP (Network B)	
on (Network A)	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?	
	Check: Is an SDP present in the 181 as a SDP answer?	
	NOTE : The presence of the P-Early-Media header in the INVITE request	
	indicates the support of "early media Authorization" in the originating	
	Network.	
	NOTE : The presence of the P-Early-Media header in the 181 indicates the	
	support of the P-Early-Media header and authorizes the media in the	
	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 → 182 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? Check: Is an SDP present in the 182 as a SDP answer? NOTE: The presence of the P-Early-Media header in the INVITE request indicates the support of "early media Authorization" in the originating Network. NOTE: The presence of the P-Early-Media header in the 182 indicates the	
	support of the P-Early-Media header and authorizes the media in the 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 if the Record-Route header is present in the INVITE establishes a communication between a user of network A and a user of network B 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	Interconnection Interface SIP (Network B)	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →	
	Apply post test routine	
Comments	Establish a communication from network A to Network B	
Comments	Check: If present 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.	
	repeat the test that an encount on a conde	

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 Il communication between a user of network A topmost header is set to the IBCF of network parameter.	and a user of network B and the
Configuration		
SIP Parameter	INVITE	
	Via: <address a="" ibcf="" in="" network="" of="">; bra</address>	nch=[any value]
Message flow SIP (Network A)	Interconnection Interface INVITE →	SIP (Network B)
	Apply post test routine	
Comments	Establish a communication from network A to Check: The topmost Via header contains the and a branch parameter. Repeat this test in reverse direction. Repeat this test with all chosen end devices.	

T(100 hard 040
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 if a Record-Route header was present in the initial INVITE 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	INVITE
	Record-Route
	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: If the Record-Route header is present is in the 180 Ringing.
	The Record-Route header is optional.
	Repeat this test in reverse direction.
	Repeat this test with all chosen end devices.

Test case number	SS_bcall_014
Test case group	BCALL/successful
Reference	5.10/ [2]
SELECTION EXPRESSION	
Test purpose	Route header in the BYE of the originating user.
	Ensure that if a Record-Route header was present in the initial INVITE the Route header may be present in the BYE request sent from the originating user equipment in network A 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="">;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: Is the Route header present in the BYE, 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 healt 045
	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 if a Record-Route header was present in the initial INVITE the Route header may be present in the BYE request sent from the terminating user equipment in network B 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: If the Route header present in the BYE, 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 if a Record-Route header was present in the initial INVITE the Route header may be present in ACK from network A upon a connection establishment from network A is completed 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)
On (Network A)	INVITE ->
	← 180 Ringing
	€ 200 OK INVITE
	ACK →
	Apply post test routine
Comments	Establish a communication from network A to Network B
	Check: Is the Route header present in the ACK, 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)</port>	
	or	
	m= Image <port number=""> Udptl or Tcptl TYPE_SDP= PIXIT (table 7.1.1-1)</port>	
	a=TYPE_SDP= PIXIT (table 1)	
	b=TYPE_SDP= PIXIT (table 1)	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
	INVITE ->	
	Apply post test routine	
Comments	Establish a communication from network A to Network B	
	Check: Is the preferred codec set to TYPE_SDP?	
	Check: If present: is the a line set to TYPE_SDP?	
	Check: If present: is the b line set to TYPE_SDP?	
	Check: Is the codec list consistent with the attribute(s) (bandwidth) regarding	
	the media description?	
	Repeat this test in reverse direction.	
	Repeat this test with all chosen end devices.	

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

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

Table 7.1.1-1

TYPE	_SDP	m= line		b= line	a= line
VA	<media></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 SIP (Network B) ACK SIP (Network B)
	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: If Fax transmission is successful? Repeat this test in reverse direction.

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

Test case number	SS_bcall_022		
Test case group	BCALL/successful		
Reference	[5] and [23]		
SELECTION EXPRESSION	[Network A] SE 45 AND [Network B] SE 45		
Test purpose	Fax transmission using the T.38 in an audio m-line codec.		
	Ensure that a Fax transmission is possible from Network A to Network B and the		
	relevant codec is the T.38 in an 'audio' m-line codec. Ensure in the active call		
	state the property of Fax transmission.		
Configuration			
SIP Parameter	INVITE: SDP		
	m=audio <port> RTP/AVP 8 OR <dynamic-pt></dynamic-pt></port>		
	a=rtpmap 8 OR <dynamic-pt> PCMA/8000</dynamic-pt>		
	m=image <port> udptl t38</port>		
	180/200 OK INVITE: SDP		
	m=image <port> udptl t38</port>		
Message flow	lutere amonties luterfees OID (Network D)		
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE (SDP1) → 180 Ringing		
	← 180 Ringing← 200 OK INVITE (SDP2)		
	ACK		
	Apply post test routine		
Comments	Establish a communication from network A to Network B		
	Check: Contains the SDP offer in the initial INVITE a T.38 codec in an 'audio'		
	line.		
	Check: Contains the SDP answer in the 180 or 200 OK INVITE a T.38 codec		
	in an 'audio' line.		
	Check: If Fax transmission is successful?		
	Repeat this test in reverse direction.		

Test case number	SS_bca	all_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		ork B] SE 4	
Test purpose	Overla	p sending, the Multiple INVITE met	hod is us	sed.
		that call establishment using overlap		
		that in the confirmed state the voice	transfer c	on the media and B-channels
	is perfo	rmed correctly.		
Configuration				
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface	_	SIP (Network B)
		INVITE(CSq 1)	→	
		INVITE(CSq 2)	→	
	←	484 Address Incomplete(CSq 1)		
		ACK	→	
		INVITE(CSq 3)	→	
	←	484 Address Incomplete(CSq 2)		
		ACK	→	
		INVITE(CSq 4)	→	
	←	484 Address Incomplete(CSq 3)	•	
	•	ACK	→	
	←	180 Ringing(CSq 4)	•	
		Apply post test routine		
Comments	Establis	sh a communication from ISDN to SIF	using th	e overlap operation in ISDN
	Check:			
	SIP ans	swers with 180 Ringing.		
		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 meth		
	Ensure that call establishment using overlap sending is performed correctled Ensure that in the confirmed state the voice transfer on the media and B-c is performed correctly.		
Configuration			
SIP Parameter	INVITE 2:		
	Supported: 100rel		
	183: Require: 100rel		
	INFO: Content-Type: application/x-session-ii SubsequentDigit: <additional digits=""></additional>	nfo	
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) → → → → →	
	180 Ringing(CSq 2) Apply post test routine		
Comments	Establish a communication from ISDN to SIP using the overlap operation in ISDN Check: All INVITE requests contains the same Call ID and From header values. Check: The 183 session Progress that establishes an early dialogue contains a Require header set to 100rel. Check: All INFO requests contain the Content-Type header set to 'application/x-session-info'. Check: All INFO requests contains the 'SubsequentDigit:' MIME body containing the additional digits. The UE B answers with 180 Ringing response after the INVITE was received. Repeat this test in reverse direction.		

Test case number	SS_bcall_025			
Test case group	BCALL/successful			
Reference	5.1.1.1.2/ [25]			
SELECTION EXPRESSION	[Network A] (SE 46 OR SE 47) AND [Network A] SE 6			
Test purpose	PSTN XML BearerCapability 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	User A is an ISDN access either in the PSTN or the SIP - ISDN interworking			
OID D	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"?></pre>			
	PSTN			
	BearerCapability			
	BCoctet3			
	CodingStandard>00<			
	InformationTransferCabability>ITC_value<			
	< BCoctet4			
	TransferMode>00<			
	InformationTransferRate>10000<			
	BCoctet5			
	Layer1Identification>01<			
	UserInfoLayer1Protocol>00011<			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE →			
0	Apply post test routine			
Comments	Check: Is a PSTN XML MIME body contained in the INVITE request? Check: If the BearerCapability element is present?			
	Check: If the InformationTransferCabability element is set as indicated in			
	table 7.1.1-1?			
	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.			
	•			

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	'01000'

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

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

Test case number	SS_bcall_028			
Test case group	BCALL/successful			
Reference	5.1.2.2/ [25]			
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6			
Test purpose	PSTN XML ProgressIndicator element in the 180.			
	User B is located in network B and an ISDN end device is used. Ensure that the			
	180 Ringing response contains a PSTN XML MIME body and at least one			
	ProgressIndicator element is present.			
Configuration	User B is an ISDN access either in the PSTN or the SIP - ISDN interworking			
	according [10] applies			
SIP Parameter	180:			
	Content-Type: application/vnd.etsi.pstn+xml			
	Content-Disposition: signal;handling=optional			
	-2vml vorsion="1.0" anading="utf.9"2>			
	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	letono anno di en letonico e			
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	J		
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			
	2 and version "1 O" enceding "utf O"2			
	<pre><?xml version="1.0" encoding="utf-8"?> PSTN</pre>			
	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	Ta			
Comments	Check: Is a PSTN XML MIME body contained in the INVITE request?			
	Check: If the first BearerCapability InformationTransferCabability element is			
	set as indicated to '00000'?			
	Check: If 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.			
<u> </u>	T			

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			
0	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			
SIP Parameter	according [10] applies			
SIP Parameter	200:			
	Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional			
	Content-Disposition: signal, nanding-optional			
	xml version="1.0" encoding="utf-8"?			
	PSTN			
	BearerCapability BCoctet3			
	CodingStandard>00<			
	InformationTransferCabability>10001<			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE ->			
	← 180 Ringing ← 200 OK INVITE			
	← 200 OK INVITE ACK →			
Apply post test routine				
Comments	Check: Is a PSTN XML MIME body contained in the 200 OK INVITE			
	response?			
	Check: Is a BearerCapability element present, the			
	InformationTransferCabability element set to '10001'?			
	Check: Is the InformationTransferCabability element value consistent with the			
	codec list in the SDP?			
	Check: Is the InformationTransferCabability element value consistent with the			
	bandwidth information in the SDP?			
	Repeat this test in reverse direction.			

Test case number	SS_bcall	_032		
Test case group	BCALL/successful			
Reference	5.1.2.3/ [25]			
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6			
Test purpose	Fall back occurs.			
	User B is located in network B and an ISDN end device is used. The Fallback			
	connection type was requested in the initial INVITE request. Ensure that the 200			
	OK INVITE response contains a PSTN XML MIME body and a BearerCapability			
	element is present the InformationTransferCabability element set to '00000'. A			
	PSTN XML MIME ProgressIndicator body is present, the ProgressDescription is set to '0000101'.			
Configuration	User B is an ISDN access either in the PSTN or the SIP - ISDN interworking			
Comigaration	according [10] applies			
SIP Parameter	200:	5 [10] 4 [20]		
	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	ı	The second secon		
SIP (Network A)		Interconnection Interface SIP (Network B)		
		INVITE →		
		180 Ringing		
	•	200 OK INVITE		
		Apply post tost routing		
Comments	Check:	Apply post test routine Is a PSTN XML MIME body contained in the 200 OK INVITE		
Comments	CHECK.	response?		
	Check:	Is a BearerCapability element present, the		
		InformationTransferCabability element set to '00000'?		
	Check:	Is a ProgressIndicator element present, the ProgressDescription is set		
		to '0000101'?		
	Check:	Is the InformationTransferCabability element value consistent with the		
		codec list in the SDP?		
	Check:	Is the InformationTransferCabability element value consistent with the		
	D · · ·	bandwidth information in the SDP?		
	Repeat tr	nis test in reverse direction.		

Test case number	SS_bcall_032A
Test case group	BCALL/successful
Reference	5.1.2.3/ [26]
SELECTION EXPRESSION	0.11.2.0/ [20]
Test purpose	Telephony events transmission
	Ensure that the ability of transmission of Telephony events can be performed by the originating user. The Telephony transmission can be done by: • either indicated in the SDP offer in the RTP stream • or SIP INFO/NOTIFY Method for DTMF tone generation
Configuration	
SIP Parameter	INVITE: CASE A m=audio <port> RTP/AVP <dynamic-pt></dynamic-pt></port>
	a=rtpmap <dynamic-pt> telephone-event/8000 a=rtpmap <dynamic-pt> 0-15 CASE B</dynamic-pt></dynamic-pt>
	Accept: application/dtmf CASE C
	Accept: application/dtmf
	NOTIFY CASE B Content-Type: application/dtmf
	'x'
	CASE C Content-Type: application/dtmf-relay
	Signal=x Duration=y
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE ->
	← 180 Ringing← 200 OK INVITEACK→
CASE A	← 200 OK INVITE
CASE A	← 200 OK INVITE ACK →
	← 200 OK INVITE ACK → RTP DTMF events INFO →
CASE B	← 200 OK INVITE ACK RTP DTMF events INFO 200 OK INFO INFO 4 200 OK INFO Apply post test routine
CASE B	← 200 OK INVITE ACK → RTP DTMF events
CASE B	RTP DTMF events INFO 200 OK INFO INFO 200 OK INFO INFO Apply post test routine Establish a communication from network A to Network B Check: Case A is the dynamic payload type 'telephone-event' present in the SDP offer? Check: Case A is the dynamic payload type 'telephone-event' covered in the RTP stream if the Telephone event occurs?
CASE B	RTP DTMF events INFO 200 OK INFO INFO 200 OK INFO Apply post test routine Establish a communication from network A to Network B Check: Case A is the dynamic payload type 'telephone-event' present in the SDP offer? Check: Case A is the dynamic payload type 'telephone-event' covered in the RTP stream if the Telephone event occurs? Check: Case B is the Content-Type header field in the INFO request conveying the DTMF signal set to 'application/dtmf'? Check: Case B contains the MIME body of the INFO request covering the
CASE B	RTP DTMF events INFO 200 OK INFO INFO 200 OK INFO Apply post test routine Establish a communication from network A to Network B Check: Case A is the dynamic payload type 'telephone-event' present in the SDP offer? Check: Case A is the dynamic payload type 'telephone-event' covered in the RTP stream if the Telephone event occurs? Check: Case B is the Content-Type header field in the INFO request conveying the DTMF signal set to 'application/dtmf'? Check: Case B contains the MIME body of the INFO request covering the TMF signal the events regarding the used content type? Check: Case C is the Content-Type header field in the INFO request
CASE B	RTP DTMF events INFO 200 OK INFO **Example 1

Test case number	SS_bcall_033
Test case group	BCALL/successful
Reference	7.1/ [24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47
Test purpose	SIP-I support, Basic call, IAM present in the INVITE request.
	Ensure that when a call initiated in the PSTN or the PLMN and the ISUP - SIP-I interworking is applicable in the originating network, an 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
	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)
Ba a a a a se di a se	[any boundary name]
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(IAM) → 100 Trying Apply post test routine
Comments	Establish a communication from network A to Network B
	Check: Is an ISUP IAM encapsulated in the INVITE request?
	Check: Are all the mandatory ISUP parameters present in the IAM and are the
	values valid?
	Check: Are the values of the optional parameters in the encapsulated IAM valid?
	Check: Is the 'm' line with corresponding attributes in the SDP consistent with the Transmission medium requirement parameter?
	Check: Is the Transmission medium requirement value consistent with the bandwidth information in the SDP?
	Repeat this test in reverse direction.

Test case number	SS_bcall_034
Test case group	BCALL/successful
Reference	7.2.1/ [24]
SELECTION EXPRESSION	[Network A] SE 4 AND SE 17 AND SE 47
Test purpose	SIP-I support, Basic call, overlap signalling.
	Ensure that when overlap signalling applies in the ISUP -SIP-I interworking in the originating network, several INVITE requests with the same Cal-ID and From tag are sent from Network A to Network B. Ensure that the original IAM is encapsulated in any INVITE request.
Configuration	
SIP Parameter	
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(1) →
	← 484 Address Incomplete(1)
	ACK INVITE(3) 484 Address Incomplete(3) ACK →
	INVITE(4) → 180 Ringing(4) Apply post test routine
Comments	Establish a communication from network A to Network B using the overlap procedure in Network A Check: Are the INVITE requests sent with the same From tag and the Call-ID? Check: After the 180 applies, are all previous INVITE transactions are
	terminated with a 484 final response? Check: Is the encapsulated IAM present in the initial INVITE request also encapsulated in any following INVITE request required for the call setup? Repeat this test in reverse direction.

Test case number	SS_bcall_035
Test case group	BCALL/successful
Reference	6.5/ [24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	SIP-I support, Basic call, ACM present in the 180 response.
	Ensure that on receipt of a 180 Ringing provisional response and an
	SIP-I - ISUP interworking is applicable in the terminating network the Backward
	call indicators parameter in the encapsulated ACM is present and the values are
	valid.
	Ensure that the values of the optional parameters in the encapsulated ACM are
	valid.
Configuration	
SIP Parameter	180:
	Content-Type: multipart/mixed;boundary=[any boundary name]
	[any houndary name]
	[any boundary name] Content-Type: application/isup;version=itu-t92
	Content-Type: application/isup,version=itu-tez 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)
0	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
	1 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2
	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
Configuration	valid. 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	[arry bourtoury numo]
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 100 Trying 183 Session Progress(ACM)
	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
	Event indicator - ALEIVING
	[any boundary name]
Message flow	[any boundary name]
SIP (Network A)	Interconnection Interface SIP (Network B)
,	INVITE -
	← 100 Trying
	← 183 Session Progress(ACM)
	← 180 Ringing(CPG)
	Apply post test routine
Comments	Establish a communication from network A to Network B
	Check: Is an ISUP CPG message encapsulated in the 180 Ringing provisional
	response?
	Check: Is the mandatory Event indicator present in the encapsulated ISUP
	CPG set to 'ALERTING'?
	Check: Are the values of optional parameters in the encapsulated ISUP CPG
	valid?
	Repeat this test in reverse direction.

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

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	[6], [7] and [6]
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) INVITE(SDP3) → 200 OK INVITE(SDP4) ACK →
CASE B	UPDATE(SDP3) 200 OK UPDATE(SDP4) Apply post test routine [5]
Comments	Establish a communication from network A to Network B using SDP1 chosen from the table 7.1.2-1
	Check: The calling user changes the media description using INVITE request containing SDP 3 codec chosen from table 7.1.2-1 different to SDP1.
	Check: Is the codec list consistent with the attribute(s) (bandwidth) regarding
	the media description? Repeat this test in reverse direction.

Test case number	SS_codec_002
Test case group	BCALL/Codec_Negotiation
Reference	[3], [4] and [5]
SELECTION EXPRESSION	
Test purpose	Session update requested by the called user.
	During the session, the called user decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re- INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. The other party sends a 200 (OK) to accept the change. The requestor responds to the 200 (OK) with an ACK. In case when the parameter in the SDP rtpmap: <dynamic-pt> is used the codecs in table 7.1.2-1 applies.</dynamic-pt>
Configuration	occoss in table 7.11.2 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)
	← 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	18 000	1		
VA_15	16	DVI4	Α	11 025	1		
VA_16	17	DVI4	Α	22 050	1		
VA_17	18	G729	Α	8 000	1		
VA_18	Dyn	G726-40	Α	8 000	1		
VA_19	Dyn	G726-32	Α	8 000	1		
VA_20	Dyn	G726-24	Α	8 000	1		
VA_21	Dyn	G726-16	Α	8 000	1		
VA_22	Dyn	G729D	Α	8 000	1		
VA_23	Dyn	G729E	Α	8 000	1		
VA_24	Dyn	GSM-EFR	Α	8 000	1		
VA_25	25	CelB	V	90 000			
VA_26	26	JPEG	V	90 000			
VA_27	28	Nv	V	90 000			
VA_28	31	H261	V	90 000			
VA_29	32	MPV	V	90 000			
VA_30	33	MP2T	V	90 000			
VA_31	34	H263	V	90 000			
VA_32	Dyn	H263-1998	V	90 000			
VA_33	Dyn	AMR	А	8 000	1		
VA_34	Dyn	AMR-WB	А	16 000	1		
VA_35	Dyn	telephone-	А	8 000	1		
		event					

7.1.3 Resource Reservation

Tast assaurantsau	100		
Test case number	SS_resource_001		
Test case group Reference	BCALL/Resource_Reservation [3], [4], [5] and [6]		
SELECTION EXPRESSION	([Network A] SE 50 AND [Network B] SE 50) AND SE 7		
Test purpose	Resource reservation successful, segmented status. Ensure that the network is able to reserve resources for quality of service when requested from the initiating user. In the INVIT the UE requests to establish QoS preconditions for all the media streams. In the 183 Session Progress the UAS supports the QoS preconditions		
	 and requests that UAC sends a confirmation when the QoS preconditions are met. The UPDATE includes in the SDP the information about the successful QoS bidirectional mode, due to the successful bidirectional PDP context established. 200 OK UPDATE the SDP contains an indication that the UE 		
0 (1 (1	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		
	OOO OK UPDATE		
	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	1. 12 F 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2		

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

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

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

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

Test case number	SS_unsucc_004	
Test case group	BCALL/unsuccessful	
Reference	[4]	
SELECTION EXPRESSION		
Test purpose	The called user is user determined busy.	
	Ensure that, when the called user is busy, the calling user with a 486 Busy Here message.	e user initiates call clearing to the
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 Netv	work 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 on	the called number.
	Ensure that when the number is char to the calling user with a 410 Gone m	nged, the network initiate call clearing nessage.
Configuration		
SIP Parameter		
Message flow		
SIP (Network A)	Interconnection Interface	SIP (Network B)
	INVITE →	
	← 410 Gone	
	ACK →	
Comments	Establish a communication from netw	vork A to Network B, user B is not
	allocated in Network B.	
	Check: Is a 410 Gone sent from N	letwork B to Network A?
	Repeat this test in reverse direction.	

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

Test case number	SS_unsucc_007		
Test case group	BCALL/unsuccessful		
Reference	[3], [4] and [5]		
SELECTION EXPRESSION			
Test purpose	Session update requested by the calling user is unsuccessful, existing		
	session remains unchanged.		
	During the session, the calling user decides to change the characteristics of the		
	media session. This is accomplished by sending a re-INVITE containing a new		
	media description. This re-INVITE references the existing dialog so that the other		
	party knows that it is to modify an existing session instead of establishing a new		
	session. Ensure that if the other party does not accept the change, he sends an		
	error response such as 488 Not Acceptable Here, which also receives an ACK. The session remains unchanged.		
Configuration	The session remains unchanged.		
SIP Parameter	INVITE: codec not supported in Network B		
Message flow	INVITE. codec not supported in Network B		
SIP (Network A) Interconnection Interface SIP (Network			
on (Notiforkin)	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.		
	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], [·] aa [o]			
Test purpose	Session update requested by the called user is unsuccessful, existing session remains unchanged.			
	During the session, the called user decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. Ensure that if the other party does not accept the change, he sends an error response such as 488 Not Acceptable Here, which also receives an ACK. The session remains unchanged. The 488 Not Acceptable Here may be sent by a simulation equipment.			
Configuration				
SIP Parameter	INVITE: codec not supported in Network A			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 180 Ringing ← 200 OK INVITE ACK →			
	Communication			
	← INVITE			
	488 Not Acceptable Here →			
	← ACK			
	Apply post test routine			
Comments	Establish a communication from network A to Network B.			
	User B in Network B attempts to change the session by sending a SDP offer to			
	the UE in Network A			
	Network A does not support the codec sent in the offer.			
	Check: Is a 488 Not Acceptable Here sent from Network B to Network A?			
	Repeat this test in reverse direction.			

Test case number	SS_unsu	cc_009		
Test case group	BCALL/unsuccessful			
Reference	[4]			
SELECTION EXPRESSION				
Test purpose	Call clea user.	ring due to no answer from	the called	user initiated by the calling
		nat when there is no answer fro all clearing to the called user		,
Configuration				
SIP Parameter				
Message flow SIP (Network A)		Interconnection Interface	_	SIP (Network B)
	←	INVITE 180 Ringing CANCEL/BYE	→ →	
	←	200 OK CANCEL/BYE		
	(487 Request Terminated ACK	→	
Comments	Check: Check: Check:	Is a CANCEL or BYE reques Is a 487 Request Terminating Are the media streams termin was sent?	g send from	n the terminating user?
	Repeat th	nis test in reverse direction.		

Test case number	SS_unsucc_010			
Test case group	BCALL/unsuccessful			
Reference	[3], [4] and [5]			
SELECTION EXPRESSION				
Test purpose	Codec not supported by the called user.			
	The initial INVITE contains a SDP with codes that does not support by the called user. Ensure that, when the called user does not accept the Media session, the called user initiate call clearing to the calling user with 488 Not Acceptable Here, which also receives an ACK.			
Configuration				
SIP Parameter	INVITE: codec not supported at user (Network B)			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) NVITE →			
CASE A	← 488 Not Acceptable Here ACK →			
CASE B	← 606 Not Acceptable ACK →			
Comments	Establish a call setup from network A to Network B. User B in Network B does not support the codec offered in the SDP received from Network A. Check: Is a 488 Not Acceptable Here sent from Network B to Network A. Repeat this test in reverse direction.			

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

Test case number	SS_unsucc_011A
Test case group	BCALL/unsuccessful
Reference	[27]
SELECTION EXPRESSION	
Test purpose	Negotiation of session timer.
	Ensure that the interconnected networks are able to negotiate the session time
	to refresh the session. If the session refresh duration is to short for one of the
	involved entities, a 422 Session Interval Too Small unsuccessful final response
	is sent in backward direction to update the session duration time. A new INVITE
Configuration	is sent and a Min-SE header present proposes a longer session duration.
Configuration Comment	The session time in Network B is smaller as the session time used in Network A
Comment	This test case is only applicable if the session refresh time is different in Network A and Network B. This situation is also load dependant.
SIP Parameter	INVITE 1:
On Tarameter	Supported: timer
	Session-Expires: x
	422:
	Min-SE. x + y
	INVITE 2
	Session-Expires: x + y
Message flow	Intersempeting Interfere
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE 1 →
	← 422 Session Interval Too Small
	ACK
	AUI
	INVITE 2 →
	← 180 Ringing
	Apply post test routine
Comments	Establish a communication setup from Network A to Network B
	Check: Is the supported header in the initial INVITE set to 'timer'
	Check: Is a 422 Session Interval Too Small send from the terminating
	Network?
	Check: Is the Session-Expires header in the second initial INVITE request sent from Network A set to the value indicated in the 422 final
	response? Repeat this test in reverse direction.
	propest the test in reverse direction.

Reference G.11.2/ [24] SELECTION EXPRESSION [Network B] SE 17 Test purpose SIP-I support. Called number is not allocated in the PSTN/PLMN network. Ensure that, when calling to an unallocated number in the PSTN/PLMN part of network B and ISUP - SIP-I interworking applies in Network B, the network initiate call clearing to the calling user with a 404 Not Found message. A ISUP REL message is encapsulated and the Cause value indicator is set to '1'. Configuration The called user number is not assigned to the PSTN/PLMN part in Network B 404: Reason: Q.850;cause=1 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] Content-Type: multipart/mixed;boundary=required REL	Test case number	SS_unsucc_012
SELECTION EXPRESSION [Network B] SE 17	Test case group	BCALL/unsuccessful
SELECTION EXPRESSION [Network B] SE 17	Reference	6.11.2/ [24]
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 SIP (Network B) INVITE 404 Not Found(REL) ACK ACK SEstablish a communication from network A to Network B, called user number is	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 SIP Parameter 404: Reason: Q.850;cause=1 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → 404 Not Found(REL) ACK → Comments Establish a communication from network A to Network B, called user number is	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 SIP Parameter 404: Reason: Q.850;cause=1 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → 404 Not Found(REL) ACK → Comments Establish a communication from network A to Network B, called user number is		
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 SIP (Network B) INVITE → 404 Not Found(REL) ACK Comments Establish a communication from network A to Network B, called user number is		
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 Comments Establish a communication from network A to Network B, called user number is		
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 Comments Establish a communication from network A to Network B, called user number is		
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) ACK → Comments Establish a communication from network A to Network B, called user number is		
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 Comments Establish a communication from network A to Network B, called user number is		Ů I
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 Comments Establish a communication from network A to Network B, called user number is	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) ACK Comments Establish a communication from network A to Network B, called user number is		
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 ACK Establish a communication from network A to Network B, called user number is		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) ACK ACK Establish a communication from network A to Network B, called user number is		
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 Comments Establish a communication from network A to Network B, called user number is		
REL Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 404 Not Found(REL) ACK Comments Establish a communication from network A to Network B, called user number is		
Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 404 Not Found(REL) ACK Comments Establish a communication from network A to Network B, called user number is		Content-Disposition: signal;nandling=required
Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 404 Not Found(REL) ACK Comments Establish a communication from network A to Network B, called user number is		REI
[any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → 404 Not Found(REL) ACK → Comments Establish a communication from network A to Network B, called user number is		1
Message flow SIP (Network A) Interconnection Interface INVITE		Gause value. 1
Message flow SIP (Network A) Interconnection Interface INVITE		[any boundary name]
INVITE → 404 Not Found(REL) ACK → Comments Establish a communication from network A to Network B, called user number is	Message flow	
← 404 Not Found(REL) ACK → Comments Establish a communication from network A to Network B, called user number is	SIP (Network A)	Interconnection Interface SIP (Network B)
ACK → Comments Establish a communication from network A to Network B, called user number is		
Comments Establish a communication from network A to Network B, called user number is		← 404 Not Found(REL)
	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 in
the Cause value of the encapsulated ISUP REL?		the Cause value of the encapsulated ISUP REL?
Repeat this test in reverse direction.		Repeat this test in reverse direction.

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 1/7'. 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] Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 [any boundary name] SIP (Network B) Message flow SIP (Network A) Interconnection Interface INVITE SIP (Network B) Message flow SIP (Network A) Interconnection Interface INVITE 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 18UP REL encapsulated and the Cause value indicator is set to	Test case number	SS_unsucc_013
SELECTION EXPRESSION [Network B] SE 17 AND SE 47	Test case group	BCALL/unsuccessful
Test purpose SIP-I support. The called user is busy. Ensure that, when the called user in the PSTN/PLMN part of Network B and ISUP - SIP-I interworking applies in Network B is busy, the network initiates call clearing to the calling user with a 486 Busy Here message. A ISUP REL message is encapsulated and the Cause value indicator is set to '17'. Configuration The called user is busy in the PSTN/PLMN part in Network B SIP Parameter 486: Reason: Q.850;cause=17 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK 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 486 Busy Here sent from Network B to Network A? Check: Is a 18UP REL encapsulated and the Cause value indicator is set to	Reference	6.11.2/ [24]
Test purpose SIP-I support. The called user is busy. Ensure that, when the called user in the PSTN/PLMN part of Network B and ISUP - SIP-I interworking applies in Network B is busy, the network initiates call clearing to the calling user with a 486 Busy Here message. A ISUP REL message is encapsulated and the Cause value indicator is set to '17'. Configuration The called user is busy in the PSTN/PLMN part in Network B SIP Parameter 486: Reason: Q.850;cause=17 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK 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 486 Busy Here sent from Network B to Network A? Check: Is a 18UP REL encapsulated and the Cause value indicator is set to	SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
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		
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		
clearing to the calling user with a 486 Busy Here message. A ISUP REL message is encapsulated and the Cause value indicator is set to '17'. Configuration The called user is busy in the PSTN/PLMN part in Network B SIP Parameter 486: Reason: Q.850;cause=17 (optional) Content-Type: multipart/mixed;boundary=[any boundary name][any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17[any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK 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 1SUP REL encapsulated and the Cause value indicator is set to		Ensure that, when the called user in the PSTN/PLMN part of Network B and
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 486: Reason: Q.850;cause=17 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK 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 1SUP REL encapsulated and the Cause value indicator is set to		
The called user is busy in the PSTN/PLMN part in Network B 486: Reason: Q.850;cause=17 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK ACK 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		
A86: Reason: Q.850;cause=17 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → 486 Busy Here(REL) ACK → ACK → ACK STORMER ACK ACK STORMER ACK ACK		
Reason: Q.850;cause=17 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface 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	Configuration	The called user is busy in the PSTN/PLMN part in Network B
Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK 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	SIP Parameter	
[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17[any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK 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 1SUP REL encapsulated and the Cause value indicator is set to		
Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK 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		Content-Type: multipart/mixed;boundary=[any boundary name]
Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK 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		
Content-Disposition: signal;handling=required REL Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK ACK SIP (Network B) 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		
REL Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK ACK SIP (Network B) 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		
Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK 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		Content-Disposition: signal;handling=required
Cause value: 17 [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK 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		DEL
[any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK ACK 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		
Message flow SIP (Network A) Interconnection Interface 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		Cause value. 17
Message flow SIP (Network A) Interconnection Interface 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		[any boundary name]
SIP (Network A) Interconnection Interface INVITE 486 Busy Here(REL) ACK 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	Message flow	any boundary namoj
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		Interconnection Interface SIP (Network B)
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	, ,	
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		← 486 Busy Here(REL)
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		ACK →
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	Comments	Establish a communication from network A to Network B, user B in the
Check: Is a ISUP RÉL encapsulated and the Cause value indicator is set to		PSTN/PLMN part of Network B is busy.
1 71/1/		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?		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) Content-Type: multipart/mixed;boundary=[any boundary name][any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 21[any boundary name]
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 480 Temporarily Unavailable (REL) ACK →
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
	SIP-I support. Call clearing due to no answer from the called user initiated
Test 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:
on raidinete.	Reason: Q.850;cause=16 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name]
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required
	REL Cause value: 16
	[any boundary name]
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 180 Ringing
CASE A	CANCEL → 200 OK CANCEL
	← 487 Request Terminated ACK →
CASE B	
	BYE(REL) → 200 OK BYE(RLC) 487 Request Terminated ACK →
Comments	Establish a communication from network A to Network B, user B does not confirm the communication. The originating user in the PSTN/PLMN part of Network A terminates the early
	dialogue. Check: Is a CANCEL or BYE request is sent from the originating network? Check: Is a ISUP REL encapsulated in a BYE request? Check: Is the Cause value of the encapsulated REL set to '16'?
	Check: If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL? Check: Is a 487 Request Terminating send from the terminating user?
	Check: Are the media streams terminated after the 200 OK CANCEL/BYE was sent? NOTE: A ISUP REL is not encapsulated in a CANCEL request.
	Repeat this test in reverse direction.

Test case number	SS_unsu	cc 016	
Test case group		nsuccessful	
Reference	7.7.1/ [24		
SELECTION EXPRESSION		A] SE 17 AND SE 47	
Test purpose	SIP-I sup by the or	port. Call clearing due to no iginating network.	answer from the called user initiated
	originating calling us interworki or BYE to	g network initiate the call cleari er is located in the PSTN/PLMI ing applies in Network A and th	confirmed by the called user, the ring after timeout of ISUP timer T9 if the IN part of Network A and ISUP - SIP-I he originating network sends a CANCEL message is encapsulated in the BYE is set to '19'.
Configuration			
SIP Parameter	Co	Q.850;cause=19 (optional) ontent-Type: multipart/mixed;bo	oundary=[any boundary name]
	Co	ontent-Type: application/isup;ve ontent-Disposition: signal;handl 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 th The ISUP Check: Check: Check: Check:	ne communication setup. It imer T9 in the PSTN/PLMN et ls a CANCEL or BYE request ls a ISUP REL encapsulated it ls the Cause value of the encapted if a Reason header is present the Cause value of the encapted.	t is sent by the originating network? in a BYE request? capsulated REL set to '19'? t, is the cause value equal to the value in osulated ISUP REL?
	Check: Check: NOTE: Repeat th		g send from the terminating user? nated after the 200 OK CANCEL/BYE ated in a CANCEL request.

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 →
	Apply post test routine
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 set to phone? Repeat this test in reverse direction. Repeat this test with all relevant end devices.

Test case number	SS_oip_002
Test case group	SIP-SIP/Service/OIP
Reference	5.2.6.3/ [2]
SELECTION EXPRESSION	
Test purpose	P-Preferred-Identity received, no match with the set of registered public identities. The terminating user receives the default public user identity of the originating user.
	In case the preconditions are fulfilled to provide the terminating UE with originating identification information without preventing the presentation, ensure that an identity information in the P-Preferred-Identity header is provided by the originating UE, does not match with the set of registered public identities of the originating UE the terminating user receives a P-Asserted-Identity based on the default public user identity associated with the originating UE identifies the
	originator of the session.
Configuration	
SIP Parameter	INVITE P-Asserted-Identity= default public user identity
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → Apply post test routine
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: If 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 relevant end 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 → Apply post test routine
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 relevant end devices.

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

Test case number	SS_oip_0	006	
Test case group	SIP-SIP/S	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-		
		eader field is derived from the Calling party number in the encapsulated	
		'Presentation restriction' indicator in the encapsulated IAM is set to	
	'allowed' ı	no Privacy value 'id' is present in the INVITE request.	
Configuration			
SIP Parameter	INVITE_		
		Asserted-Identity=[derived from the ISUP calling party number]	
	Co	ontent-Type: multipart/mixed;boundary=[any boundary name]	
		any boundary name]	
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
		IAM	
	Calling party number		
	Screening indicator		
	Network provided or user provided, verified and		
	passed		
	Presentation restriction [24]		
	allowed		
	Address signal		
	[a	any boundary name]	
Message flow			
SIP (Network A)		Interconnection Interface SIP (Network B)	
		INVITE(IAM) →	
		Apply post test routine	
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	
	Chaole	'allowed'?	
	Check:	Is the P-Asserted-Identity header field derived from the Calling party	
	Check:	number in the encapsulated IAM? Is the value 'id' not present in the Privacy header field (if included)?	
		is the value of not present in the Privacy header field (if included)?	
	rkepeattn	iis test iii reverse direction.	

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

7.1.5.2 Test purposes for OIR

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

Test case number	SS_oir_002
Test case group	SIP-SIP/Service/OIR
Reference	4.3.2, 4.5.2.4/ [7]
SELECTION EXPRESSION	SE 20 AND SE 25
Test purpose	Communication forwarding unconditional, served user subscribes OIR.
	The user A and user C are in network B and user C is provided with OIP. The user B is in network A and is provided with CFU "diverting number is released to the diverted-to user" = Yes. In case the served user subscribes Originating Identification Restriction (e.g. permanent mode), ensure that when user A calls user B, the call is forwarded unconditional to user C, user C is not informed of the forwarding number. The diverted-to user receives no identity of the diverting user neither in a History-Info header nor in the To header.
Configuration	Diverting user subscribes to the OIR service
SIP Parameter	INVITE1: no history entry present INVITE2: History-Info header: <sip:userb@networka?privacy=history>;index=1, <sip: userc@networkb;cause="302">;index=1.1</sip:></sip:userb@networka?privacy=history>
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	← INVITE1
	CFU is performed in Network A INVITE2 →
	Apply post test routine
Comments	Check: No History-Info header is received in the INVITE from Network B. Check: Is the Privacy value history is escaped in the hi-targed-to-uri of the diverting user in Network A? Repeat this test in reverse direction.
	Repeat this test with all chosen end devices.

Test case number	SS_oir_003	
Test case group	SIP-SIP/Service/OIR	
Reference	7.1.3/ [24]	
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 52	
Test purpose	SIP-I support. ISUP Calling party number presentation restricted in the	
	encapsulated IAM.	
	Ensure when BICC/ISUP - SIP-I interworking applies in the originating network	
	the BICC/ISUP IAM is encapsulated in the INVITE request. The	
	P-Asserted-Identity header field is derived from the Calling party number in the	
	encapsulated IAM. The 'Presentation restriction' indicator in the encapsulated IAM is set to 'restricted' the value 'id' is present in the Privacy header of the	
	INVITE request.	
Configuration	INVITE request.	
SIP Parameter	INVITE	
On Tarameter	P-Asserted-Identity=[derived from the ISUP calling party number]	
	Privacy: id	
	Content-Type: multipart/mixed;boundary=[any boundary name]	
	[any boundary name]	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	IAM	
	Calling party number	
	Screening indicator	
	Network provided or user provided, verified and	
	passed	
	Presentation restriction	
	restricted	
	Address signal	
	[any boundary name]	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
on (Network A)	INVITE(IAM) →	
	Apply post test routine	
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	
	[any boundary name]	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(IAM) → Apply post test routine	
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]		
SELECTION EXPRESSION				
Test purpose	Originat	ing user receives the identity of t	he termi	nating user.
	terminati originatir	n case the preconditions are fulfilleding identification information without ng UE receives in a 1xx or 200 SIP in eld with a valid public user identity of	preventi response	ng the presentation , the e a P-Asserted-Identity
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	← 2	00 OK INVITE(P-Asserted-Identity) Apply post test routine		
Comments		Is the P-Asserted-Identity is prese Progress or in a 200 OK INVITE? his test in reverse direction. his 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
O and in constitute	was received.
Configuration SIP Parameter	Special arrangement for the terminating user exists
SIP Parameter	INVITE
	Supported: from-change
	200 OK INVITE
	Supported: from-change
	P-Asserted-Identity:
	, and the second
	UPDATE
	From: (second user identity)
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE →
	← 180 Ringing ← 200 OK INVITE(P-Asserted-Identity)
	← 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 present in a 180 Ringing or 183 Session
	Progress or in a 200 OK INVITE?
	Check: Is the 'from-change' tag present in the supported header of the
	provisional (18x) or final (200 OK) response?
	Check: Is an UPDATE request sent by the terminating user containing a From
	header field set to the value send by the terminating user?
	Repeat this test in reverse direction.
	Repeat this test with all chosen end devices.

Test case number	SS_tip_003		
Test case group	SIP-SIP/Service/TIP		
Reference	4.5.2.9/ [8]		
SELECTION EXPRESSION	SE 21 AND SE 22 AND [Network B] SE 48		
Test purpose	Second identity not provided.		
	Ensure that, when the option tag "from-change" in the Supported header field		
	is provided by the originating UE in the INVITE request, the terminating user		
	does not receive the from-change tag in the initial INVITE, no from-change tag is		
	sent in the 200 OK INVITE response, an UPDATE containing a second identity is		
	sent and the From header is set to the default public user identity of the		
Configuration	terminating user.		
Configuration SIP Parameter	Special arrangement for the terminating user does not exist		
SIP Parameter	INVITE Supported: from-change		
	Supported. Hom-change		
	200 OK INVITE		
	P-Asserted-Identity:		
	, i		
	UPDATE		
	From: (default public user identity)		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B) INVITF →		
	=		
	← 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 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		
	7		
	Connected number		
	Screening indicator Notwork provided or user provided verified and		
	Network provided or user provided, verified and passed		
	Address presentation restriction		
	allowed		
	Address signal		
	, 188. 333 3.g. 18.		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(IAM) →		
	← 180 Ringing(ACM)		
	← 200 OK INVITE(ANM)		
	ACK →		
	Apply post test routine		
Comments	Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final		
	response?		
	Check: Is the Screening indicator in the encapsulated ANM set to 'Network		
	provided' or 'user provided, verified and passed'?		
	Check: Is the Address presentation restriction indicator in the encapsulated ANM set to allowed?		
	Repeat this test in reverse direction.		
L	וויפף במנ נוווס נפטנ ווו ופייפוספ עוופטנוטוו.		

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 encapsulated 200 OK. Ensure that on receipt of a 200 OK INVITE to establish a confirmed dialogue ANM is encapsulated if SIP-I - BICC/ISUP interworking is applicable in Networking. A Generic number parameter is present the Number qualifier indicator set '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	an ork to		
Configuration	'allowed'. The terminating user in the PSTN/PLMN part of Network B is subscribed to the COLP 'no screening option'.	ne		
SIP Parameter	COLP 'no screening option' 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			
Magaza day	[any boundary name]			
Message flow SIP (Network A)	Interconnection Interface 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 provide not verified'?Check: Is the Address presentation restriction indicator in the Generic num			
	set to 'allowed'? Repeat this test in reverse direction.	IDGI		
	propositing test in reverse direction.			

7.1.5.4 Test purposes for TIR

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

Test case number	SS_tir_001A		
Test case group	SIP-SIP/Service/TIR		
Reference	4.5.2.6.2.2/[9]		
SELECTION EXPRESSION	SE 23		
Test purpose	CDIV occurs. Originating user does not receive the identity of the served user.		
	Ensure that, when Call diversion occurs, the identity of the CDIV served user is restricted when the CDIV served user is subscribed to the TIR service and requires to prevent the presentation of his/here identity. The hi-entry of the History-Info header in the 181 identifying the served user contains an escaped 'Privacy' header set to 'history'.		
Configuration	The served user subscribes to the 'TIR' service		
SIP Parameter	181 History-Info header: <sip:userb@networkb?Privacy=history>;index=1, <sip: userc@networkb;cause="[any]">;index=1.1</sip:></sip:userb@networkb?		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 181 Being Forwarded INVITE		
	Apply post test routine		
Comments	Check: Is the History-Info header present in the 181 sent to the originating user?		
	Check: Is the Privacy header is escaped in the hi-entry identify the served user set to 'history'?		
	Repeat this test in reverse direction.		
	Repeat this test with all chosen end devices.		

Test case number	SS_tir_0	01B		
Test case group	SIP-SIP/Service/TIR			
Reference	4.5.2.7/ [9]			
SELECTION EXPRESSION	SE 23	1		
Test purpose	CDIV oc	curs. Originating user does no	t receive the identity of the served	
	user.	0 0	,	
	Ensure th	nat, when Call diversion occurs, t	he identity of the diverted-to user is	
	restricted	I when the diverted-to user is sub	oscribed to the TIR service and requires	
		nt the presentation of his/here ide		
			the 180 or 200 OK INVITE identifying	
		ted-to user contains an escaped		
Configuration	The diverted-to user subscribes to the 'TIR' service			
SIP Parameter	180/200	OK		
	History-Info header:			
	<sip:userb@networkb>;index=1,</sip:userb@networkb>			
	<sip: userc@networkb;cause="[any]?<b">Privacy=history>;index=1.1</sip:>			
Message flow				
SIP (Network A)		Interconnection Interface	SIP (Network B)	
	_	INVITE(1)	→	
	←	INVITE(2)	_	
	180 Ringing(2) →			
	←	180 Ringing(1)	_	
		200 OK INVITE(2)	→	
	(ACK		
	←	200 OK INVITE(1)		
		ACK	→	
	01	Apply post test routine	.: 11 400 000 014 11 11	
Comments	Check:		ent in the 180 or 200 OK sent to the	
	Chaok	originating user?	d in the hi entry identify the diverted to	
	Check:		d in the hi-entry identify the diverted-to	
	Donoct #	user set to 'history'?		
	Repeat this test in reverse direction.			
	repeat th	nis test with all chosen end devic	es.	

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 encar	psulated 200 OK.				
		at on receipt of a 200 OK INVITE to establish a confirmed dialogue an				
		ncapsulated if SIP-I - BICC/ISUP interworking is applicable in Network				
		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 IN					
	Co	ontent-Type: multipart/mixed;boundary=[any boundary name]				
	_					
		any boundary name]				
		ontent-Type: application/isup;version=itu-t92				
		Content-Disposition: signal;handling=required				
		ANM				
		Connected number				
		Screening indicator				
		Network provided or user provided, verified and				
		passed				
		Address presentation restriction				
		restricted				
	Address signal					
		· ·				
	[any boundary name]					
Message flow						
SIP (Network A)		Interconnection Interface SIP (Network B)				
	_	INVITE(IAM) →				
		← 180 Ringing(ACM)				
	← 200 OK INVITE(ANM)					
		ACK →				
		Apply post test routine				
Comments	Check:	Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final				
	Cheek	response?				
	Check:	Check: Is the Screening indicator in the encapsulated ANM set to 'Network				
	Chack	provided' or 'user provided, verified and passed'?				
	Check:	Check: Is the Address presentation restriction indicator in the encapsulated ANM set to allowed?				
	Reneat th	nis test in reverse direction.				
	Inepeat th	113 LEST 111 LEVELSE MILECTION.				

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			
Configuration	'restricted'. 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			
Magaza flow	[any boundary name]			
Message flow SIP (Network A)	Interconnection Interface INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM) ACK Apply post test routine SIP (Network B) → APPLY NOTE: APPLY N			
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_0	001	
Test case group	SIP-SIP/S	ervice/HOLD	
Reference	4.5.2.1/[1	7]	
SELECTION EXPRESSION	SE 24		
Test purpose	Hold the	session the media stream was p	reviously set to sendrecv.
	UPDATE i attribute "a		s done containing the SDP with the esting the hold session <i>receives</i> 200
Configuration		•	
SIP Parameter			
Message flow SIP (Network A)	A co	Interconnection Interface nfirmed session already exists	SIP (Network B)
CASE A	←	INVITE(<mark>sendonly</mark>) 200 OK INVITE (recvonly) ACK	→ →
CASE B	←	UPDATE(sendonly) 200 OK UPDATE (recvonly) Apply post test routine	→
Comments	Check: Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? Repeat this test in reverse direction.		

Test case group SIP-SIP/Service/HOLD Reference 4.5.2.1/ [17] SELECTION EXPRESSION SE 24				
SELECTION EXPRESSION SE 24				
SELECTION EXPRESSION SE 24				
Test purpose Hold the session the media stream was previously set to recvonly.				
Ensure that the UE B requesting hold of the session stops sending media and	Ensure that the UE B 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=sendonly".	containing the SDP with the attribute "a=sendonly".			
	The UE A after requesting to hold the held session sends an INVITE or			
UPDATE request containing the SDP with the attribute "a=inactive".				
Configuration				
SIP Parameter				
Message flow				
SIP (Network A) Interconnection Interface SIP (Network B)				
A confirmed session already exists				
CASE A				
200 OK INVITE (recvonly) →				
← ACK				
INVITE (<mark>inactive</mark>) →				
← 200 OK INVITE (inactive)				
ACK →				
CASE B ← INVITE (sendonly)				
200 OK INVITE (recvonly) →				
← ACK				
UPDATE(inactive) →				
← 200 OK UPDATE (inactive)				
CASE C ← UPDATE (sendonly)				
200 OK UPDATE (recvonly) →				
INVITE (inactive) →				
200 OK INVITE (inactive)				
ACK →				
NON 2				
CASE D ← UPDATE (sendonly)				
200 OK UPDATE (recvonly) →				
UPDATE(inactive) →				
← 200 OK UPDATE (inactive)				
Apply post test routine				
Comments Check: Is the user in network B able to set the session on hold by sending	an			
INVITE or UPDATE request and the version parameter in the SDP				
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				
line is incremented?				
Repeat this test in reverse direction.				

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

Test case number	SS_hold_	004	
Test case group	SIP-SIP/Service/HOLD		
Reference			
	4.5.2.1/[17]		
SELECTION EXPRESSION	SE 24		
Test purpose	Resume the session the media stream was previously set to inactive.		
	The Operation is to the Winnerthall Co. E		
		sion is in the "inactive" state. Ensure	
	resume the session with user B the UE-A sends an INVITE or UPDATE to		
	resume the session with the attribute "a=recvonly in the SDP. The UE A after requesting to resume the held session <i>receives</i> 200 OK final response with the		
			es 200 OK final response with the
	attribute	<mark>'a=sendonly</mark> in the SDP.	
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	→
CASE B	←	INVITE(sendonly)	
		200 OK INVITE (recvonly)	→
	←	ACK	
		UPDATE(<mark>inactive</mark>)	→
	←	200 OK UPDATE (inactive)	
		INVITE (recvonly)	→
	←	200 OK (sendonly)	
		ACK	→
CASE C	←	UPDATE (sendonly)	
		200 OK UPDATE (recvonly)	→
		INVITE(<mark>inactive</mark>)	→
	←	200 OK INVITE (inactive)	
		ACK	→
		UPDATE (<mark>recvonly</mark>)	→
	←	200 OK UPDATE (<mark>sendonly</mark>)	
CASE D	←	UPDATE (sendonly)	
		200 OK UPDATE (recvonly)	→
		UPDATE(<mark>inactive</mark>)	→
	←	200 OK UPDATE (inactive)	
		UPDATE (<mark>recvonly</mark>)	→
	←	200 OK UPDATE (<mark>sendonly</mark>)	
		Apply post test routine	
Comments	Check:		the session on hold by sending an
		INVITE or UPDATE request and the	e version parameter in the SDP 'o'
		line is incremented?	
	Check:		the session on hold by sending an
		INVITE or UPDATE request and th	e version parameter in the SDP 'o'
		line is incremented?	
	Check:	Is the user in network A able to reti	
		INVITE or UPDATE request and th	e version parameter in the SDP 'o'
		line is incremented?	
	Repeat th	nis 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 B sends an INVITE or UPDATE request to hold the session.
	Hold is done containing the SDP with the attribute "a=sendonly". The UE A
	sends a 200 OK final response containing the SDP with the attribute
	"a=recvonly" and stops sending media.
Configuration	
SIP Parameter	
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	A confirmed session already exists
CASE A	← INVITE(sendonly)
	200 OK INVITE(recvonly) →
	← ACK
CASE B	← UPDATE(sendonly)
CASE B	200 OK UPDATE (recvonly) →
	Apply post test routine
Comments	Check: Is the user in network B able to set the session on hold by sending an
Comments	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	SIP-SIP/Service/HOLD			
Reference	4.5.2.1/[17]			
SELECTION EXPRESSION	SE 24			
Test purpose	Hold the session the media stream was previously set to sendonly.			
	The Session is in the held state done by UE-A. Ensure that the UE B 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 receiving the hold request sends 200 OK final response containing the SDP with the attribute "a=inactive"			
		s sending media.	SDP with the attribute a=mactive	
Configuration	and stop	s sending media.		
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 (inactive)		
	_	200 OK INVITE (inactive)	→	
	←	ACK		
CASE B		INVITE(sendonly)	→	
CASE B	←	200 OK INVITE (recvonly)	7	
	•	ACK	→	
	←	UPDATE (inactive)	-	
		200 OK UPDATE (inactive)	→	
		,		
CASE C	_	UPDATE (sendonly)	→	
	(200 OK UPDATE (recvonly)		
	←	INVITE (inactive)	→	
	←	200 OK INVITE (<mark>inactive</mark>) ACK	7	
		ACK		
CASE D		UPDATE (<mark>sendonly</mark>)	→	
	←	200 OK UPDATE (recvonly)		
	←	UPDATE (inactive)		
		200 OK UPDATE (<mark>inactive</mark>)	→	
_	T	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'			
	Charles	line is incremented?	at the ecosion on held by conding an	
	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?			
	Reneat t	nis test in reverse direction.		
<u> </u>	Ivehear r	no toot iii leveloe uliectioil.		

Test case number	SS_hold	_007	
Test case group	SIP-SIP/S	Service/HOLD	
Reference	4.5.2.1/[17]	
SELECTION EXPRESSION	SE 24	-	
Test purpose	Resume	the session the media stream wa	as previously set to recvonly.
	resume to containing Resume the attrib	Ensure that the UE B sends an INVITE or UPDATE request requesting to resume the session with user A, the UE-B starts sending media. Resume is done containing the SDP with the attribute "a=sendrecv". The UE A after receiving the Resume of the session sends 200 OK final response containing the SDP with the attribute "a=sendrecv". The a=sendrecv attribute is the default value therefore the attribute can be omitted.	
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
		onfirmed session already exists	
CASE A	←	INVITE (sendonly)	
		200 OK INVITE(<mark>recvonly</mark>)	→
	←	ACK	
	←	INVITE(sendrecv)	
		200 OK INVITE(<mark>sendrecv</mark>)	→
	←	ACK	
CASE B	←	UPDATE (sendonly)	
		200 OK UPDATE (recvonly)	→
	←	UPDATE (sendrecv)	
		200 OK UPDATE (sendrecv)	→
		Apply post test routine	
Comments	Check:	Is the user in network B able to se	et the session on hold by sending an
		INVITE or UPDATE request and the	he version parameter in the SDP 'o'
		line is incremented?	
	Check:	Is the user in network B able to re-	trieve the session by sending an
		INVITE or UPDATE request and the	he version parameter in the SDP 'o'
		line is incremented?	-
	Repeat th	nis test in reverse direction.	

Test case number	SS_hold_	008	
Test case group		Service/HOLD	
Reference	4.5.2.1/ [17]		
SELECTION EXPRESSION	SE 24	17]	
		the session the media stream wa	s proviously set to inactive
Test purpose	Resume	the session the media stream wa	is previously set to mactive.
	The See	sion is in the "inactive" state. Ensure	that the LIE B conde on INVITE or
		request requesting to resume the s	
		media. Resume is done containing the	
			esume of the session sends 200 OK
0	final resp	onse containing the SDP with the at	ttribute a= <mark>sendonly</mark> .
Configuration			
SIP Parameter			
Message flow			a
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 (<mark>sendonly</mark>)	→
	←	ACK	
CASE B	_	INVITE (sendonly)	→
	←	200 OK INVITE (recvonly)	_
	_	ACK	→
	←	UPDATE (<mark>inactive</mark>)	_
	_	200 OK UPDATE (inactive)	→
	←	UPDATE (<mark>recvonly</mark>)	_
		200 OK UPDATE (<mark>sendonly</mark>)	→
		LIBBATE (
CASE C	_	UPDATE (sendonly)	→
	(200 OK UPDATE (recvonly)	
	←	INVITE (inactive)	_
	-	200 OK INVITE (inactive)	→
	÷	ACK	
	←	INVITE (recvonly)	_
	-	200 OK INVITE (sendonly)	7
	←	ACK	
CASE D		LIDDATE (see deals)	
CASE D	_	UPDATE (sendonly)	→
	-	200 OK UPDATE (recvonly)	
	~	UPDATE (inactive)	_
	_	200 OK UPDATE (inactive)	→
	←	UPDATE (recvonly)	→
		200 OK UPDATE (sendonly)	7
Commonto	Chasky	Apply post test routine	t the coording on held by conding on
Comments	Check:		t the session on hold by sending an
		•	ne version parameter in the SDP 'o'
	Chook	line is incremented?	t the session on hold by conding an
	Check:		t the session on hold by sending an
			ne version parameter in the SDP 'o'
	Chaste	line is incremented?	rious the appaier by ser director
	Check:	Is the user in network B able to ret	
			ne version parameter in the SDP 'o'
	Dono-4 "	line is incremented?	
<u> </u>	repeat th	nis test in reverse direction.	

Test case number	SS_hold_009	
Test case group	SIP-SIP/Service/HOLD	
Reference	4.5.2.1/ [17]	
SELECTION EXPRESSION	SE 24	
	Resume the session on both sides the med	dia etroam was proviously set to
Test purpose	inactive.	dia stream was previously set to
	The Considering the Uline of the United States	that the LIE A is as supplied to
	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 ses	ssion with the attribute
	"a= sendonly in the SDP.	
	The UE A after requests to resume the sessio	
	containing the SDP with the attribute "a=recve	
	The UE B after requests to resume the sessio	
	request containing the SDP with the attribute attribute is the default value therefore the attri	
Message flow	attribute to the default value therefore the attri	outo dan so dimitida.
SIP (Network A)	Interconnection Interface	SIP (Network B)
On (Notwork 74)	A confirmed session already exists	on (Notwork B)
CASE A	INVITE(sendonly)	→
OAGE A	← 200 OK INVITE (recvonly)	
	ACK	→
	← INVITE(inactive)	
	200 OK INVITE (inactive)	→
	← ACK	
	INVITE(sendonly)	→
	€ 200 OK INVITE (recvonly)	
	ACK	→
	← INVITE(sendrecv)	
	200 OK INVITE (sendrecv)	→
	← ACK	
	AON	
CASE B	INVITE(sendonly)	→
OAGE B	← 200 OK INVITE (recvonly)	
	ACK	→
	← UPDATE (inactive)	
	200 OK UPDATE (inactive)	→
	INVITE(sendonly)	→
	€ 200 OK INVITE (recvonly)	
	ACK	→
	← UPDATE (sendrecv)	-
	200 OK UPDATE (sendrecv)	→
	(coa.co.)	-
CASE C	UPDATE (sendonly)	→
	← 200 OK UPDATE (recvonly)	
	← INVITE(inactive)	
	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 (<mark>sendonly</mark>)	→
	← 200 OK UPDATE (recvonly)	
	← UPDATE (sendrecv)	
	200 OK UPDATE (sendrecv)	→
	Apply post test routine	
•	· · • ·	

Comments	Check:	Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?
	Check:	Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?
	Check:	Is the user in network A able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?
	Check:	Is the user in network B able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? The absence of the 'sendrecv' attribute is the default value.
	Repeat tl	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 me	edia stream was previously set to
Took purpose	inactive.	Jaia Strouin was previously set to
	macuve.	
	The Session is in the "inactive" state. Ensure	that the LIF R sends an INIVITE or
	UPDATE request to resume the session with	
	media. Resume is done containing the SDP	
	The UE A after receiving the Resume of the	
	response containing the SDP with the attribu	
	The UE A after requests to resume the sessi	
	request containing the SDP with the attribute	
	The UE B after receiving the Resume of the	
	response containing the SDP with the attribu	
	attribute is the default value therefore the att	ribute can be omitted.
Configuration		
SIP Parameter		
Message flow		
SIP (Network A)	Interconnection Interface	SIP (Network B)
	A confirmed session already exists	
CASE A	INVITE(sendonly)	
	200 OK INVITE (recvonly)	→
	← ACK	
	INVITE(inactive)	→
	← 200 OK INVITE (inactive)	
	ACK	→
	← INVITE(sendonly)	
	200 OK INVITE (recvonly)	→
	← ACK	
	INVITE(sendrecv)	→
	← 200 OK INVITE (sendrecv)	
	ACK `	→
CASE B	← INVITE(sendonly)	
	200 OK INVITE (recvonly)	→
	← ACK `	
	UPDATE (inactive)	→
	← 200 OK UPDATE (inactive)	
	← INVITE(sendonly)	
	200 OK INVITE (recvonly)	→
	← ACK	_
	UPDATE (<mark>sendrecv</mark>)	→
	← 200 OK UPDATE (sendrecv)	-
	200 OR OF BATE (Scharcov)	
CASE C	← UPDATE (sendonly)	
07.02 0	200 OK UPDATE (recvonly)	→
	INVITE(inactive)	→
	← 200 OK INVITE (inactive)	
	ACK	→
	← UPDATE (sendonly)	7
	200 OK UPDATE (recvonly)	→
		→
	INVITE(sendrecv) € 200 OK INVITE (sendrecv)	7
	,	_
	ACK	→
CASED	LIDDATE (condents)	
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	
	1171	

Comments	Check:	Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?
	Check:	Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?
	Check:	Is the user in network B able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?
	Check:	Is the user in network A able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? The absence of the 'sendrecv' attribute is the default value.
	Repeat t	his test in reverse direction.

—	00 1 11 044	
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	INVITE	
	Content-Type: multipart/mixed;boundary=[any boundary name]	
	[any boundary name]	
	a=sendonly	
	[any boundary name]	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	CPG	
	Generic notification	
	remote hold	
	[any boundary name]	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
	A confirmed session already exists	
CASE A	INVITE(<mark>sendonly</mark> , CPG <mark>hold</mark>) →	
	← 200 OK INVITE (recvonly)	
	ACK →	
	Apply post test routine	
Comments	Establish a session from Network A to Network B	
	The user in the PSTN/PLMN part of Network A places the session on hold.	
	Check: Is a CPG encapsulated in the INVITE request?	
	Check: Is a Generic notification parameter present the Notification indicator	
	set to 'remote hold'?	
	Check: Is the 'a' attribute in the SDP set to 'sendonly'?	
	Check: Is the Version parameter in the SDP incremented?	
	Repeat this test in reverse direction.	

Test case number	SS_hold_012		
Test case group	SIP-SIP/Service/HOLD		
Reference	B.10/ [24]		
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 54		
Test purpose	SIP-I support. Hold requested by the called user.		
	Ensure that when an INVITE request updates a confirmed session a CPG is		
	encapsulated if SIP-I - ISUP interworking is applicable in Network B. The		
	Generic Notification Indicator parameter is present set to 'hold'. The 'a' attribute		
	is set to 'sendonly' present in the SDP.		
	In the 200 OK INVITE the 'a' attribute is set to 'recvonly' present in the SDP.		
Configuration			
SIP Parameter	INVITE:		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	a-condenly		
	a=sendonly		
	[any boundary name]		
	[any boundary name] Content-Type: application/isup;version=itu-t92		
	Content-Type: application/isup,version=itu-i92 Content-Disposition: signal;handling=required		
	20 2		
	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		
Comments	Establish a session from Network A to Network B The user in the PSTN/PLMN part of Network B places the session on hold.		
	Check: Is a CPG encapsulated in the INVITE request?		
	Check: Is a Generic notification parameter present the Notification indicator		
	set to 'remote hold'?		
	Check: Is the 'a' attribute in the SDP set to 'sendonly'?		
	Check: Is the Version parameter in the SDP incremented?		
	Repeat this test in reverse direction.		
	I reposit the test in reverse direction		

Test case number	SS_hold_013		
Test case group	SIP-SIP/Service/HOLD		
Reference	B.10/ [24]		
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 54		
Test purpose	SIP-I support. Hold requested by the originating user, Hold by the terminating user. Retrieve requested by the originating user.		
	Ensure the hold and retrieve procedure when ISUP - SIP-I interworking applies in the Network A:		
	 Originating user in Network A places the session on hold. Terminating user in Network B places the session on hold. Originating user in Network A retrieves the session. Terminating user in Network B retrieves the session. 		
	Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A.		
Configuration			
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic notification remote hold		
	or remote retrieval		
	[any boundary name]		
Message flow SIP (Network A) CASE A	Interconnection Interface A confirmed session already exists INVITE(sendonly, CPG hold) ← 200 OK INVITE (recvonly) ACK		
	 ← INVITE(inactive) 200 OK INVITE (inactive) ← ACK 		
	INVITE(sendonly, CPG retrieval) → 200 OK INVITE (recvonly) ACK →		
	← INVITE(sendrecv) 200 OK INVITE (sendrecv) →		
	← ACK Apply post test routine		
Comments	Establish a session from Network A to Network B		
	The user in the PSTN/PLMN part of Network A places the session on hold.		
	Check: Is a CPG encapsulated in the INVITE request? Check: Is a Generic notification parameter present the Notification indicator set to 'remote hold'?		
	Check: Is the 'a' attribute in the SDP set to 'sendonly'? Check: Is the Version parameter in the SDP incremented?		
	The user in Network B places the session on hold		
	Check: Is the 'a' attribute in the SDP set to 'inactive'?		
	Check: Is the Version parameter in the SDP incremented?		
	The user in Network A retrieves the session Check: Is a CPG encapsulated in the INVITE request?		
	Check: Is a Generic notification parameter present the Notification indicator		
	set to 'remote retrieval'? Check: Is the 'a' attribute in the SDP set to 'sendonly'?		
	Check: Is the Version parameter in the SDP incremented?		
	The user in Network B retrieves the session		
	Check: Is the 'a' attribute in the SDP set to 'sendrecv'?		
	Check: Is the Version parameter in the SDP incremented?		
	Repeat this test in reverse direction.		

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

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 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	INVITE TOQUOC HOTH the Network 7 t.	
SIP Parameter	INVITE:	
on rarameter	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 A confirmed session already exists INVITE(sendonly) 200 OK INVITE (recvonly) → ACK	
	INVITE(inactive, CPG hold) 200 OK INVITE (inactive) ACK →	
	ACI	
	INVITE(recvonly, CPG retrieval) → 200 OK INVITE (sendonly)	
	ACK →	
	← INVITE(sendrecv)	
	200 OK INVITE (sendrecv) →	
	← ACK	
	Apply post test routine	
Comments	Establish a session from Network A to Network B	
	The user in Network B places the session on hold.	
	Check: Is the 'a' attribute in the SDP set to 'sendonly'?	
	Check: Is the Version parameter in the SDP incremented?	
	The user in Network A places the session on hold	
	Check: Is a CPG encapsulated in the INVITE request? Check: Is a Generic notification parameter present the Notification indicator	
	set to 'remote hold'? Check: Is the 'a' attribute in the SDP set to 'inactive'?	
	Check: Is the Version parameter in the SDP incremented?	
	The user in Network A retrieves the session	
	Check: Is a CPG encapsulated in the INVITE request?	
	Check: Is a Generic notification parameter present the Notification indicator set to 'remote retrieval'?	
	Check: Is the 'a' attribute in the SDP set to 'recvonly'?	
	Check: Is the Version parameter in the SDP incremented?	
	The user in Network B retrieves the session	
	Check: Is the 'a' attribute in the SDP set to 'sendrecv'?	
	Check: Is the Version parameter in the SDP incremented?	
	Repeat this test in reverse direction.	

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		
Configuration	INVITE request from the Network A.		
SIP Parameter	INVITE:		
oir raidilletei	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 (recvonly) →		
	← ACK		
	INVITE(<mark>inactive</mark> , CPG <mark>hold</mark>) →		
	200 OK INVITE (inactive)		
	ACK →		
	← INVITE(<mark>sendonly</mark>)		
	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 Chack: Is a CDC encount of the INVITE request?		
	Check: Is a CPG encapsulated in the INVITE request? Check: Is a Generic notification parameter present the Notification indicator		
	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 Version parameter in the SDP incremented?		
	Repeat this test in reverse direction.		
	The France of Section Section 2015		

7.1.5.6 Communication Diversion (CDIV)

7.1.5.6.1 Communication Forwarding Unconditional (CFU)

Test case number	SS_cfu_0	001		
Test case group	SIP-SIP/S	Service/CFU		
Reference	4.5.2.6/ [9	9]		
SELECTION EXPRESSION	SE 25			
Test purpose	Commur	nication forwarding uncondition	al, basic r	ules.
	The user A and user C are in Network A. The user B is in network B and is provided with CFU. Ensure that when user A calls user B, the call is forwarded unconditional to user			
	C. In the	active call state, ensure the prope	rty of spee	ch.
Configuration				
SIP Parameter				
Message flow SIP (Network A)	← ← ←	Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) ACK(Call-ID B-C) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) Communication Apply post test routine	→ → →	SIP (Network B)
Comments	Check: CDIV unconditional is successful. Check: In the active call state, ensure the property of speech. Check: Is the P-Asserted-Identity present set to the identity of the originating user? Repeat this test in reverse direction.			

Test case number	SS_cfu_002	
Test case group	SIP-SIP/Service/CFU	
Reference	4.5.2.6/ [9]	
SELECTION EXPRESSION	SE 25 AND SE 30	
Test purpose	Communication forwarding unconditional, no notification.	
	The user A and user C are in Network A. The user B is in network B and is provided with CFU, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B, the call is forwarded unconditional to user C, the originating user is not notified.	
Configuration	Subscription options:	
	Originating user receives notification that his communication has been diverted = No	
SIP Parameter		
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFU is performed	
	← INVITE(Call-ID B-C)	
	180 Ringing(Call-ID C-B) →	
	← 180 Ringing(Call-ID B-A)	
	Apply post test routine	
Comments	Check: No notification regarding call forwarding in network B is received at the interconnection interface.	
	Repeat this test in reverse direction.	

Test case number	SS_cfu_003	
Test case group	SIP-SIP/Service/CFU	
Reference	4.5.2.6/ [9]	
SELECTION EXPRESSION	SE 25 AND SE 30	
Test purpose	Communication forwarding unconditional, originating user is notified. URI of the diverted-to user not received.	
	The user A and user C are in network A. The user B is in network B and is provided with CFU "Originating user receives notification that his communication has been diverted" = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and not informed of the diverted-to number and served user number.	
Configuration	Subscription options:	
	 Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user 	
	in diversion notification = No	
	 Served user allows the presentation of his/her URI to originating user in diversion notification = No 	
SIP Parameter	181 Being Forwarded	
	History-Info:	
	<pre><sip:userb@networkb?privacy=history>;index=1,</sip:userb@networkb?privacy=history></pre>	
	<sip: userc@networka;cause="302<mark">?Privacy=history>;index=1.1</sip:>	
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)	
	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	
Comments	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>;index=1, <sip: userc@networka;cause="302">;index=1.1</sip:></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 number	SS_cfu_006		
Test case group	SIP-SIP/Service/CFU		
Reference	4.5.2.6/ [9]		
SELECTION EXPRESSION	SE 25 AND SE 30		
Test purpose	Communication forwarding unconditional, diverted-to user receives the		
	URI of the served user.		
	The user A and user C are in network A. The user B is in network B and is		
	provided with CFU "Served user allows the presentation of his/her URI to		
	diverted-to user" = Yes.		
	Ensure that when user A calls user B, the call is forwarded unconditional to user		
	C, user C is informed of the forwarding number.		
Configuration	Subscription options:		
	 Served user allows the presentation of his/her URI to diverted-to user = Yes 		
SIP Parameter	INVITE:		
	Request line contains ';cause=302'		
	History-Info header:		
	<sip:userb@networkb>;index=1,</sip:userb@networkb>		
"	<sip: userc@networka;cause="302">;index=1.1</sip:>		
Message flow	laterance (in laterance)		
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B) →		
	CFU is performed MVITF(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 '302'?		
	Check: Is the cause parameter in the last entry is set to '302'?		
	NOTE: The history entries can be accumulated in "one" History-Info header or		
	each history entry is present in one single History-Info header.		
	Repeat this test in reverse direction.		
	• •		

Test case number	SS_cfu_007	
Test case group	SIP-SIP/Service/CFU	
Reference	4.5.2.6/[9]	
SELECTION EXPRESSION	SE 25 AND SE 30	
Test purpose	Communication forwarding unconditional, full notification.	
	The user A and user C are in network A. The user B is in network B and is provided with CFU Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, and "Served user allows the presentation of his/her URI to diverted-to user" = Yes.	
	Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and informed of the diverted-to number and	
	user C is informed of the forwarding number.	
Configuration	Subscription options:	
	Originating user receives notification that his communication has been	
	diverted = <mark>Yes</mark>	
	Served user allows the presentation of forwarded to URI to originating user	
	in diversion notification = Yes	
CID Devements:	• 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:userb@networkb>	
	<sip: userc@networka;cause="302">;index=1.1</sip:>	
	Colp. 00010 @ network t,000000=0022,1100x=1.1	
	181 Being Forwarded	
	History-Info header:	
	<sip:userb@networkb>;index=1,</sip:userb@networkb>	
	<sip: userc@networka;cause="408">;index=1.1</sip:>	
	200 OK INVITE	
	History-Info header:	
	<sip:userb@networkb>;index=1,</sip:userb@networkb>	
	<sip: userc@networka;cause="486">;index=1.1</sip:>	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
	INVITE(Call-ID A-B) →	
	CFU is performed	
	 ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A 	
	180 Ringing(Call-ID C-B)	
	← 180 Ringing(Call-ID B-A)	
	200 OK INVITE(Call-ID C-B) →	
	← ACK(Call-ID C-B)	
	← 200 OK INVITE(Call-ID B-A)	
	ACK(Call-ID A-B) →	
	Communication	
	Apply post test routine	
Comments	Check: A History-Info header is received in the INVITE at the interconnection	
	interface sent to user C containing the URI identifying the served user.	
	Check: A History-Info header is received in the 181 Being Forwarded at the	
	interconnection interface sent to user A containing the URI identifying	
	the diverted-to user.	
	Check: Is the 'cause' parameter present in the Request line sent to user C	
	(diverted-to user) set to '302'?	
	NOTE: The history entries can be accumulated in "one" History-Info header or	
	each history entry is present in one single History-Info header.	
	Repeat this test in reverse direction.	

Test case number	SS_cfu_008		
Test case group	SIP-SIP/Service/CFU		
Reference	4.5.2.6/ [9]		
SELECTION EXPRESSION	SE 25		
Test purpose	Communication forwarding unconditional, unsuccessful UDUB.		
	The user A and user C are in network A. The user B is in network B and is provided with CFU. Ensure that when user A calls user B, the call is forwarded unconditional to user C user C is user determined user busy.		
Configuration	·		
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B) →		
	CFU is performed		
	► INVITE(Call-ID B-C)		
	486 Busy Here(Call-ID C-B) →		
	← ACK(Call-ID B-C)		
	← 486 Busy Here(Call-ID A-B)		
	ACK(Call-ID A-B) →		
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here.		
	Repeat this test in reverse direction.		

Test case number	SS_cfu_009		
Test case group	SIP-SIP/Service/CFU		
Reference	1.5.2.6/ [9]		
SELECTION EXPRESSION	SE 25		
Test purpose	Communication forwarding unconditional, unsuccessful NDUB.		
	The user A and user C are in network A. The user B is in network B. Ensure that when user A calls user B, the call is forwarded uncondition and user C is network determined user busy.	onal to user	
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface SIP (Netwo	ork B)	
	INVITE(Call-ID A-B) →		
	CFU is performed		
	← INVITE(Call-ID B-C)		
	486 Busy Here(Call-ID C-B) →		
	← ACK(Call-ID B-C)		
	← 486 Busy Here(Call-ID A-B)		
	ACK(Call-ID A-B) →		
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here.		
	Repeat this test in reverse direction.		

Test case number	SS_cfu_010	
Test case group	SIP-SIP/Service/CFU	
Reference	4.5.2.6/ [9]	
SELECTION EXPRESSION	SE 25 AND SE 30 AND [Network A] SE 9	
Test purpose	Communication forwarding unconditional, interaction with a not trusted network.	
	The user A and user C are in network A. The user B is in network B and is provided with CFU Originating user receives notification that his communication has been diverted = Yes "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = 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 user C is not informed of the forwarding number.	
Configuration	Subscription options:	
	 Originating user receives notification that his communication has been diverted = Yes 	
	 Served user allows the presentation of forwarded to URI to originating user in diversion notification = No 	
	 Served user allows the presentation of his/her URI to originating user in 	
	diversion notification = No	
	 Served user allows the presentation of his/her URI to the diverted-to user = No 	
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →	
	CFU is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A)	
	Apply post test routine	
Comments	Check: No History-Info header is received in the INVITE at the	
	interconnection interface.	
	Check: No History-Info header is received in the 181 Being Forwarded at the interconnection interface (if sent).	
	Repeat this test in reverse direction.	

Toot coop number	00 atv. 044
Test case number	SS_cfu_011
Test case group	SIP-SIP/Service/CFU
Reference	6.5/ [24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	SIP-I support. CFU performed in Network B, Notification subscription options is set to presentation not allowed.
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user
	number. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in the
	Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options:
_	 Calling user receives notification that his call has been diverted (forwarded or deflected) = no
SIP Parameter	183 Session Progress
	Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name]
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ACM
	Backward call indicator
	Called party's status indicator
	no indication
	Redirection number
	Address signal (<i>Diverted-to user</i>)
	Call diversion information
	Notification subscription options presentation not allowed
	Redirecting reason
	unconditional
	Generic notification
	call is diverting
	San is an orang
	[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)
	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
l'est purpose	options is set to presentation allowed without redirection number.
	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
	LOU
	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
	unconditional
	Generic notification
	call is diverting
	, and the second
	[any boundary name]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE(Call-ID A-B) →
	CFU is performed
	← INVITE(Call-ID B-C, IAM)
	← 183 Session Progress (Call-ID B-A, ACM)
	Apply post test routine
Comments	Originating user in Network A establishes a call to user in Network B. Network B
	performs the diversion to a user in Network A
	Check: 183 Session Progress is received at the interconnection interface.
	Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'?
	Check: Is the 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.
	rropout this test in reverse uncollon.

Test sees number	00 -6: 040
Test case number	SS_cfu_013
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 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.
Configuration	Subscription options:
	 Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number
SIP Parameter	183 Session Progress
	Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name]
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ACM
	Backward call indicator
	Called party's status indicator
	no indication
	Redirection number
	Address signal (<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 haundam nama]
Message flow	[any boundary name]
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →
	CFU is performed
•	INVITE(Call-ID B-C, IAM)
·	183 Session Progress (Call-ID B-A, ACM)
	Apply post test routine
Comments	Originating user in Network A establishes a call to user in Network B. Network B
	performs the diversion to a user in Network A
	Check: 183 Session Progress is received at the interconnection interface.
	Check: Is an ACM encapsulated in the 183?
	Check: Is the Called party's status indicator set to 'no indication'?
	Check: Is the Redirection number present?
	Check: Is Notification subscription options indicator set to 'presentation
	allowed with redirection number'?
	Check Is the Redirecting reason set to 'unconditional'?
	Repeat this test in reverse direction.

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
l set pui pees	number.
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part
	of Network B and is provided with CFU, Diverted-to user is subscribed to the
	COLR service in Permanent mode.
	Ensure that when user A calls user B, the call is forwarded unconditional to user
	C, a Redirection number restriction parameter is present set to 'Presentation
	restricted' in the encapsulated ANM contained in the 200 OK INVITE if
	ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	Subscription options:
	 Connected user subscribed to COLR, Permanent = yes
SIP Parameter	200 OK
	Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name]
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ANM
	Redirection number restriction
	Presentation restricted
	Flesentation restricted
	[any boundary name]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE(Call-ID A-B), IAM →
	CFU is performed
•	← INVITE(Call-ID B-C)
	180 Ringing (Call-ID C-B, ACM) →
•	+ 180 Ringing (Call-ID B-A)
	200 OK INVITE (Call-ID C-B, ANM) →
	ACK (Call-ID B-C)
•	200 OK INVITE (Call-ID B-A)
	ACK (Call-ID A-B) →
	Apply post test routine
Comments	Originating user in Network A establishes a call to user in Network B. Network B
	performs the diversion to a user in Network A
	Check: Is a 200 OK INVITE received at the interconnection interface?
	Check: Is an ANM encapsulated in the 200 OK? Check: Is an ANM encapsulated in the 200 OK?
	Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation
	restricted'?
	Repeat this test in reverse direction.

Test case number	SS_cfu_015
Test case group	SIP-SIP/Service/CFU
Reference	6.7/ [24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	SIP-I support. CFU performed in Network B, No restriction of the
	Redirection number.
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part
	of Network B and is provided with CFU, Diverted-to user is not subscribed to the
	COLR service.
	Ensure that when user A calls user B, the call is forwarded unconditional to user
	C, if a Redirection number restriction parameter is present it is set to
	'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE
	if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	Subscription options:
	 Connected user subscribed to COLR = no
SIP Parameter	200 OK
	Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name]
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ANM
	Redirection number restriction
	Presentation allowed
	or
	Redirection number restriction not present
	Troumbot Tootholl That produit
	[any boundary name]
Message flow	<u> </u>
SIP (Network A)	Interconnection Interface SIP (Network B)
, ,	INVITE(Call-ID A-B), IAM →
	CFU is performed
	← INVITE(Call-ID B-C)
	180 Ringing (Call-ID C-B, ACM) →
	+ 180 Ringing (Call-ID B-A)
	200 OK INVITE (Call-ID C-B, ANM) →
	ACK (Call-ID B-C)
•	← 200 OK INVITE (Call-ID B-A)
	ACK (Call-ID A-B) →
	Apply post test routine
Comments	Originating user in Network A establishes a call to user in Network B. Network B
	performs the diversion to a user in Network A
	Check: Is a 200 OK INVITE received at the interconnection interface?
	Check: Is an ANM encapsulated in the 200 OK?
	Check: Is the ISUP/BICC Redirection number restriction present set to
	'Presentation allowed' or is the parameter absent?
	Repeat this test in reverse direction.

Test case number	SS_cfu_016
Test case group	SIP-SIP/Service/CFU
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	SIP-I support. CFU performed in Network B, Notification of diverted-to user
	Redirecting number 'presentation allowed'.
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part
	of Network B and is provided with CFU, Served user releases his/her number to
	diverted-to user = Release diverting number information.
	Ensure that when user A calls user B, the call is forwarded unconditional to user
	C, user C is notified of call diversion and informed of the diverting number.
	The notification information is present in the encapsulated IAM contained in the
	Redirecting number 'presentation allowed' and Redirection information if
	ISUP/BICC - SIP-I interworking is applicable in Network B.
Configuration	Subscription options:
	• Served user releases his/her number to diverted-to user = Release diverting
	number information
SIP Parameter	INVITE
	Content-Type: multipart/mixed;boundary=[any boundary name]
	[any have dam unama]
	[any boundary name]
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required
	Content-Disposition. Signal, nanding=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
	unconditional
	[any boundary name]
Message flow	OID (N. c. J. D)
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE(Call-ID A-B) →
	CFU is performed NVITE(Call-ID B-C. IAM)
1	(
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
	'unconditional'?
	Repeat this test in reverse direction.
	1 -1

SIP-SIP/Service/CFU	Test case number	SS_cfu_017				
SELECTION EXPRESSION Network B SE 17 AND SE 47 AND SE 55 SIP-I support. CFU performed in Network B, Notification of diverted-to use 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 use 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. Subscription options: • Served user releases his/her number to diverted-to user = Do not release diverting number/information 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 Redirecting information Original Redirection Reason unknown Redirecting indicator Redirecting indicator Redirecting indicator Redirecting indicator Redirecting indicator Redirecting indicator Redirecting reason unconditional [any boundary name]- Comments Originating user in Network A establishes a call to user in Network B, Network B, Nattre Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B, SiP (Network B) enforms the diversion to a user in Network A intercenced at the Address presentation restricted or section restricted? Check: Is a INVITE (call-ID A-B) presentation restricted? Check: Is the Original called number present and the Address pres						
Network B SE 17 AND SE 47 AND SE 55						
SIP-I support. CFU performed in Network B, Notification of diverted-to use 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 use 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 S. SIP-I interworking is applicable in Network B. Subscription options: Subscription options: Seved user releases his/her number to diverted-to user = Do not release diverting number information INVIE* Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/sup;version=itu-t92 Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name]- Wessage flow SiP (Network A) Interconnection interface INVITE(Call-ID A-B) CPU is performed INVITE(Call-ID B-C, IAM) Apply post test routine CPU is performed INVITE(Call-ID B-C, IAM) Apply post test routine CPU is performed INVITE(Call-ID B-C, IAM) Apply post test routine CPU is performed INVITE(Call-ID B-C, IAM) Apply post test routine CPU is an IAM encapsulated in the INVITE(Call-ID B-C, IAM) Apply post test rou						
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 use 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. Subscription options: Subscription options: Served user releases his/her number to diverted-to user = Do not release diverting numberinformation INVITE* Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/sup;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 sedirection counter Redirection counter Redirection counter Redirection counter Redirection restricted indicator is set to presentation restricted? Cipal is an IAM encapsulated in the INVITE? Check: Is an IAM encapsulated in the INVITE? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirection number present and the Address presentation restricted indicator is		[Network B] SE 17 AND SE 47 AND SE 55				
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. Subscription options: ■ Served user releases his/her number to diverted-to user = Do not release diverting numberinformation 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-t92 Content-Type: application restricted indicator presentation restricted and calculation and presentation restricted indicator Redirection information Original Redirection information Original Redirection restricted Redirection Redirection Presentation restricted Preforms the diversion to a user in Network A stablishes a call to user in Network B. Network B. Preforms the diversion to a user in Network A Check: Is all NITE (Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network B. Siperforms the diversion to a user in Network A check: Is all NITE request received at the interconnection interface? Check: Is the Redirecting number present and the Address presentation restricted? Check: Is the Redirectinn present of present	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				
Subscription options: Served user releases his/her number to diverted-to user = Do not release diverting numberinformation 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 signal (Diverting user) Original Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting indicator Redirecting indicator Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E performs the diversion to a user in Network A. Check: Is a INVITE request received at the interconnection interface? Check: Is a INVITE request received at the interconnection restricted? Check: Is the Redirection number present and the Address presentation restricted? Check: Is the Redirection number present and the Address presentation restricted? Check: Is the Redirection number present and the Address presentation restricted? Check: Is the Redirection number present and the Address presentation restricted? Check: Is the Redirection number present and the Address presentation restricted? Check: Is Redirection number present and the Address presentation restricted? Check: Is Redirection number present and the Address presentation restricted? Check: Is Redirection number present and the Address presentation restricted? Check: Is Redirection number present and the Address presentation restricted? Check: Is Redirection number present and the Address presentation restricted? Check: Is Redirection number present and the Address presentation restricted? Check: Is Redirection number present and the Address presentation restricted		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				
Served user releases his/her number to diverted-to user = Do not release diverting numberinformation Notification Notification	Configuration					
Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM	oogaranon	• Served user releases his/her number to diverted-to user = Do not release				
[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM	SIP Parameter	INVITE				
Redirecting number Address presentation restricted indicator presentation restricted Address signal (Diverting user) Original called number Address presentation restricted indicator presentation restricted indicator presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason unconditional[any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E performs the diversion to a user in Network A Check: Is an IAM encapsulated in the INVITE? 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 Redirection number present and the Address presentation restricted indicator is set to 'presentation restricted'? Check: Is the Redirection number present and the Redirecting reason is set to 'presentation restricted'? Check: Is the Redirection number present and the Redirecting reason is set to 'presentation restricted'? Check: Is the Redirection information present and the Redirecting reason is set to 'presentation restricted'?		[any boundary name] Content-Type: application/isup;version=itu-t92				
Redirecting number Address presentation restricted indicator presentation restricted Address signal (Diverting user) Original called number Address presentation restricted indicator presentation restricted indicator presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason unconditional[any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E performs the diversion to a user in Network A Check: Is an IAM encapsulated in the INVITE? 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 Redirection number present and the Address presentation restricted indicator is set to 'presentation restricted'? Check: Is the Redirection number present and the Redirecting reason is set to 'presentation restricted'? Check: Is the Redirection number present and the Redirecting reason is set to 'presentation restricted'? Check: Is the Redirection information present and the Redirecting reason is set to 'presentation restricted'?						
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 indicator Redirecting reason unconditional[any boundary name] Wessage flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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 Redirection number present and the Redirecting reason is set to 'Check: Is the Redirection information present?						
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[any boundary name] Wessage flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network E performs the diversion to a user in Network A Check: Is an IAM encapsulated in the INVITE? 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 Redirection number present and the Redirecting reason is set to Check: Is the Redirection number present? Check: Is Redirection information present? Check: Is Redirection information present and the Redirecting reason is set to						
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[any boundary name] Wessage flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E performs the diversion to a user in Network A 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 and the Redirecting reason is set to 'presentation restricted'? Check: Is the Redirection number present? Check: Is the Redirection information present and the Redirecting reason is set to 'presentation restricted'? Check: Is the Redirection information present and the Redirecting reason is set to 'presentation restricted'? Check: Is Redirection information present and the Redirecting reason is set to 'presentation restricted'? Check: Is Redirection information present and the Redirecting reason is set to 'presentation restricted'? Check: Is Redirection information present and the Redirecting reason is set to 'presentation restricted'? Check: Is Redirection information present and the Redirecting reason is set to 'presentation restricted'? Check: Is Redirection information present and the Redirecting reason is set to 'presentation restricted'? Check: Is Redirection information present and the Redirecting reason is set to 'presentation restricted'?						
Original called number Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network E performs the diversion to a user in Network A 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 Redirection number present and the Address presentation restricted indicator is set to 'presentation restricted'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation restricted'? Check: Is the Redirection number present? Check: Is the Redirection present? Check: Is Redirection information present and the Redirecting reason is set to						
Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason unconditional[any boundary name] Wessage flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E performs the diversion to a user in Network A Check: Is a INVITE request received at the interconnection interface? 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						
Presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason unconditional[any boundary name] Wessage flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed NVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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 number present and the Redirecting reason is set to 'presentation restricted'? Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to		Original called number				
Presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason unconditional[any boundary name] Wessage flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed NVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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 number present and the Redirecting reason is set to 'presentation restricted'? Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to		presentation restricted Address signal Redirection information				
Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason unconditional[any boundary name] Wessage flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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 number present? Check: Is Redirection number present?						
Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason unconditional[any boundary name] Wessage flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network E performs the diversion to a user in Network A Check: Is a INVITE request received at the interconnection interface? 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						
unknown Redirecting indicator Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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 and the Address presentation restricted indicator is set to 'presentation restricted'? Check: Is the Redirection number present and the Redirection'? Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to						
Redirection counter Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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						
Redirection counter Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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						
Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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						
unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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						
[any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) 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: 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						
Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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		direction				
SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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		[any boundary name]				
INVITE(Call-ID A-B) CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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		Interconnection Interface OID (Nature 1-D)				
CFU is performed NVITE (Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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	SIP (Network A)					
Apply post test routine Originating user in Network A establishes a call to user in Network B. Network E 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						
Originating user in Network A establishes a call to user in Network B. Network E 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						
Originating user in Network A establishes a call to user in Network B. Network E 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	•					
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	Comments					
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	Comments					
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						
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						
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						
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						
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						
Check: Is the Redirection number present?Check: Is Redirection information present and the Redirecting reason is set to						
Check: Is Redirection information present and the Redirecting reason is set to						
المسممة المنافعة المن						
		'unconditional'?				
Repeat this test in reverse direction.		Repeat this test in reverse direction.				

7.1.5.6.2 Communication Forwarding Busy (CFB)

-	1			
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	Commu	Communication forwarding busy, basic rules.		
	The user	A and user C are in Network A. T	he user B i	s in network B and is
	provided	with CFB.		
	Ensure th	hat when user A calls user B, the o	call is forwa	arded busy to user C. In the
	active ca	Il state, ensure the property of spe	ech.	
Configuration				
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CFB is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		
Comments	Check:	CDIV busy is successful.		
	Check:	In the active call state, ensure the		
	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_0	002		
Test case group		Service/CFB		
Reference	4.5.2.6/ [9]		
SELECTION EXPRESSION	SE 26 A	ND SE 30		
Test purpose	Commu	nication forwarding busy, no n	otification.	
	provided that his c Ensure th	A and user C are in Network A. with CFB, subscription option: "(ommunication has been diverted that when user A calls user B, the g user is not notified.	Driginating υ d" = No.	ser receives notification
Configuration	Subscrip	otion options:		
	 Orig 	nating user receives notification	that his con	nmunication has been
	dive	<mark>rted</mark> = <mark>No</mark>		
SIP Parameter				
Message flow SIP (Network A)		Interconnection Interface INVITE(Call-ID A-B)	→	SIP (Network 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 fo	rwarding in	network B is received at the
		interconnection interface.		
	Repeat tl	nis 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		
l set pui pees	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:		
, and the second	 Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = No 		
	 Served user allows the presentation of his/her URI to originating user in 		
	diversion notification = No		
SIP Parameter	181 Being Forwarded <sip:userb@networkb?privacy=history&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486?privacy=history>;index=1.1</sip:userc@networka;cause=486?privacy=history></sip:userb@networkb?privacy=history&reason=sip;cause=486>		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed INVITE(Call-ID B-C)		
	← 181 Being Forwarded (Call-ID B-A)		
	180 Ringing(Call-ID C-B) →		
	← 180 Ringing(Call-ID B-A)		
	Apply post test routine		
Comments	Check: A 181 Being Forwarded and a History-Info header is received at the		
	interconnection interface in both entries in the History-Info header a		
	Privacy header is escaped value 'history'.		
	Check: Is the cause parameter in the last entry set to '486'?		
	NOTE: The history entries can be accumulated in "one" History-Info header or		
	each history entry is present in one single History-Info header.		
	Repeat this test in reverse direction.		

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

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		
rest purpose	of the served user.		
	of the served dser.		
	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 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:userb@networkb?privacy=history&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)		
	Apply post test routine		
Comments	Check: A History-Info header 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	DID (No. 1 D)			
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-C) Apply post test routine			
Comments	Check: A History-Info header received in the INVITE contains the URI of user			
Comments	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.			
L				

Test case number	SS_cfb_007
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/ [9]
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	Communication forwarding busy, full notification.
	The user A and user C are in network A. The user B is in network B and is
	provided with CFB "Originating user receives notification that his communication
	has been diverted" = Yes "Served user allows the presentation of forwarded to
	URI to originating user in diversion notification" = Yes, "diverting number is
	released to the diverted-to user" = Yes.
	Ensure that when user A calls user B, the call is forwarded busy to user C, user
	A is notified of call diversion and informed of the diverted-to number and user C
	is informed of the forwarding number.
Configuration	Subscription options:
	 Originating user receives notification that his communication has been
	diverted = Yes
	 Served user allows the presentation of forwarded to URI to originating user
	in diversion notification = Yes,
	 diverting number is released to the diverted-to user = Yes
SIP Parameter	INVITE:
	Request line contains ';cause=486'
	History-Info header:
	<pre><sip:userb@networkb&reason=sip;cause=486>;index=1,</sip:userb@networkb&reason=sip;cause=486></pre>
	<pre><sip: userc@networka;cause="486">;index=1.1</sip:></pre>
	Top: door of notwork speaked 1007 shindow 111
	181 Being Forwarded
	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:>
	Colp. 40010 ⊕1101W0110 1,044000 = 100≥,11140X=1.11
	200 OK INVITE
	History-Info header:
	<pre><sip:userb@networkb&reason=sip;cause=486>;index=1,</sip:userb@networkb&reason=sip;cause=486></pre>
Message flow	<sip: userc@networka;cause="486">;index=1.1</sip:>
Message flow SIP (Network A)	<sip: userc@networka;cause="486">;index=1.1</sip:>
Message flow SIP (Network A)	<pre><sip: userc@networka;cause="486">;index=1.1</sip:></pre> Interconnection Interface SIP (Network B)
	<pre><sip: userc@networka;cause="486">;index=1.1 Interconnection Interface</sip:></pre>
	<pre> <sip: userc@networka;cause="486">;index=1.1 Interconnection Interface</sip:></pre>
	<pre></pre>
	<pre></pre>
	<pre>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)</pre>
	<pre>Interconnection Interface</pre>
	<pre></pre>
	csip: userC@networkA;cause=486>;index=1.1 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 B-A) 200 OK INVITE(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → CFB is performed SIP (Network B) SIP (Network B) ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → CFB is performed INVITE(Call-ID C-B) → ACK(Call-ID C-B) → CFB is performed INVITE(Call-ID C-B) → ACK(Call-ID C-B) → CFB is performed INVITE(Call-ID C-B) → ACK(Call-ID C-B) → CFB is performed INVITE(Call-ID C-B) → CFB is performed CFB is performed INVITE(Call-ID C-B) → CFB is performed INVITE(Call-ID C-B) → CFB is performed CFB is performed CFB is performed CFB is performed INVITE(Call-ID C-B) → CFB is performed CFB is perfo
	Interconnection Interface SIP (Network B)
	Interconnection Interface SIP (Network B)
	Interconnection Interface SIP (Network B)
SIP (Network A)	Interconnection Interface SIP (Network B)
	Interconnection Interface SIP (Network B)
SIP (Network A)	Interconnection Interface SIP (Network B)
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB 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) Communication Apply post test routine 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
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB 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) Communication Apply post test routine 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
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed INVITE(Call-ID B-C) I81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) ACK(Call-ID C-B) Communication Apply post test routine 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.
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 C-B) → 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ACK(Call-ID C-B) Communication Apply post test routine 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. Check: Is the 'cause' parameter present in the Request line sent to user C
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed INVITE(Call-ID B-C) 181 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ACK(Call-ID C-B) Communication Apply post test routine 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. Check: Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '486'?
SIP (Network A)	Interconnection Interface INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID C-B) INVITE(Call-ID C-B) INVITE(Call-ID C-B) ACK(Call-ID C-B) Communication Apply post test routine 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'?
SIP (Network A)	Interconnection Interface INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID B-A) 180 Ringing(Call-ID B-A) 180 Ringing(Call-ID C-B) 180 Ringing(Call-ID C-B) ACK(Call-ID C-B) Communication Apply post test routine 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 C 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
SIP (Network A)	Interconnection Interface INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID C-B) INVITE(Call-ID C-B) INVITE(Call-ID C-B) ACK(Call-ID C-B) Communication Apply post test routine 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'?

SS_cfb_0	008		
SIP-SIP/	Service/CFB		
4.5.2.6/ [9]		
SE 26			
Commu	nication forwarding busy, unsuc	cessful U	DUB.
	•		
The user	A and user C are in network A. T	he user B is	s in network B and is
provided	with CFB.		
1.		call is forwa	arded busy to user C and
			,
	,		
	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-ÌD B-C)		
←	486 Busy Here(Call-ID A-B)		
	ACK(Call-ID A-B)	→	
Check:	The dialogue is terminated by re	eceiving a 4	86 Busy Here.
Repeat th	nis test in reverse direction.		•
	SIP-SIP/S 4.5.2.6/ [I SE 26 Commun The user provided Ensure th user C is	Communication forwarding busy, unsuce The user A and user C are in network A. T provided with CFB. Ensure that when user A calls user B, the cuser C is user determined user busy. Interconnection Interface 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)	SIP-SIP/Service/CFB 4.5.2.6/ [9] SE 26 Communication forwarding busy, unsuccessful U The user A and user C are in network A. The user B is provided with CFB. Ensure that when user A calls user B, the call is forward user C is user determined user busy. Interconnection Interface INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C) 486 Busy Here(Call-ID C-B) CK(Call-ID B-C) 486 Busy Here(Call-ID A-B) ACK(Call-ID A-B) ACK(Call-ID A-B) Check: The dialogue is terminated by receiving a 44

Test case number	SS_cfb_	009		
Test case group		Service/CFB		
Reference	4.5.2.6/			
		9]		
SELECTION EXPRESSION	SE 26			
Test purpose	Commu	nication forwarding busy, unsuc	ccessful N	DUB.
			. 5.	
		A and user C are in network A. T	he user B i	s in network B and is
		with CFB.		
		hat when user A calls user B, the	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-ID A-B)	→	
Comments	Check:	A 181 Being Forwarded is recei	ved at netv	vork 1 originating access.
	Check:	The dialogue is terminated by re	eceiving a	186 Busy Here.
	Repeat t	his test in reverse direction.	Ü	-

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" = No, "diverting number is released to the diverted-to user" = 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 user C is not informed of the forwarding number.		
Configuration	Subscription options:		
	 Originating user receives notification that his communication has been 		
	diverted = <mark>Yes</mark>		
	 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 No		
SIP Parameter	INVITE: no History-Info header		
	181 Being Forwarded no History-Info header		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B) →		
	CFB is performed		
	★ INVITE(Call-ID B-C) ★ 181 Being Forwarded(Call-ID B-A)		
	g		
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 (if sent).		
	Repeat this test in reverse direction.		
L	proposit tino tost in foroido direction.		

Test case number	SS_cfb_011
Test case group	SIP-SIP/Service/CFB
Reference	6.5/ [24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	SIP-I support. CFB performed in Network B, Notification subscription options is set to presentation not allowed.
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on busy user to user
	C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
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 (<i>Diverted-to user</i>) Call diversion information Notification subscription options presentation not allowed 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
	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 User Busy'? Repeat this test in reverse direction.

Toot coop number	CC -th 040	
Test case number	SS_cfb_012	
Test case group	SIP-SIP/Service/CFB	
Reference	6.5/ [24] [Network B] SE 17 AND SE 47 AND SE 55	
SELECTION EXPRESSION		
Test purpose	SIP-I support. CFB performed in Network B, Notification subscription options is set to presentation allowed without redirection number.	
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number.	
	Ensure that when user A calls user B, the call is forwarded on busy user to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.	
Configuration	Subscription options:	
_	 Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number 	
SIP Parameter	183 Session Progress	
	Content-Type: multipart/mixed;boundary=[any boundary name]	
	[any boundary name]	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	ACM	
	Backward call indicator	
	Called party's status indicator	
	no indication	
	Redirection number	
	Address signal (<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		
SIP (Network A)	Interconnection Interface SIP (Network B)	
	INVITE(Call-ID A-B) →	
	CFB is performed	
	← INVITE(Call-ID B-C, IAM)	
•	← 183 Session Progress (Call-ID B-A, ACM)	
	Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B	
	performs the diversion to a user in Network A	
	Check: 183 Session Progress is received at the interconnection interface.	
	Check: Is an ACM encapsulated in the 183?	
	Check: Is the Called party's status indicator set to 'no indication'?	
	Check: Is the Redirection number present?	
	Check: Is Notification subscription options indicator is set to 'presentation	
	allowed without redirection number'?	
	Check: Is the Redirecting reason set to 'User Busy'?	
	Repeat this test in reverse direction.	

Toot ooo number	00 ath 040		
Test case number	SS_cfb_013		
Test case group	SIP-SIP/Service/CFB		
Reference	6.5/[24]		
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55		
Test purpose	SIP-I support. CFB performed in Network B, Notification subscription options is set to presentation allowed with redirection number.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number.		
	Ensure that when user A calls user B, the call is forwarded on busy user to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.		
Configuration	Subscription options:		
garanen.	 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 (<i>Diverted-to user</i>)		
	Address signal		
	Call diversion information		
	Notification subscription options		
	presentation allowed with redirection number		
	Redirecting reason		
	User Busy		
	Generic notification		
	call is diverting		
	[any baymalan mana]		
Message flow	[any boundary 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 Notification subscription options indicator is set to 'presentation		
	allowed with redirection number'?		
	Check: Is the Redirecting reason set to 'User Busy'?		
	Repeat this test in reverse direction.		

Test case number	SS_cfb_014		
Test case group	SIP-SIP/Service/CFB		
Reference	6.7/ [24]		
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53		
Test purpose	SIP-I support. CFB performed in Network B, Restriction of the Redirection		
	number.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part		
	of Network B and is provided with CFB, Diverted-to user is subscribed to the		
	COLR service in Permanent mode.		
	Ensure that when user A calls user B, the call is forwarded on busy user to user		
	C, a Redirection number restriction parameter is present set to 'Presentation		
	restricted' in the encapsulated ANM contained in the 200 OK INVITE if		
Configuration	ISUP/BICC- SIP-I interworking is applicable in Network A.		
Configuration	Subscription options:		
SIP Parameter	 Connected user subscribed to COLR, Permanent = yes 200 OK 		
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		
	Content-Disposition. Signal, nationing=required		
	ANM		
	Redirection number restriction		
	Presentation restricted		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B), IAM →		
	CFB is performed		
•	INVITE(Call-ID B-C)		
	180 Ringing (Call-ID C-B, ACM) → 180 Ringing (Call-ID B-A)		
`	200 OK INVITE (Call-ID C-B, ANM) →		
	ACK (Call-ID B-C)		
	£ 200 OK INVITE (Call-ID B-A)		
	ACK (Call-ID A-B) →		
	Apply post test routine		
Comments	Originating user in Network A establishes a call to user in Network B. Network B		
	performs the diversion to a user in Network A		
	Check: Is a 200 OK INVITE received at the interconnection interface?		
	Check: Is an ANM encapsulated in the 200 OK?		
	Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation		
	restricted'?		
	Repeat this test in reverse direction.		

Test case number	SS_cfb_015		
Test case group	SIP-SIP/Service/CFB		
Reference	6.7/ [24]		
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53		
Test purpose	SIP-I support. CFB performed in Network B, No restriction of the		
	Redirection number.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part		
	of Network B and is provided with CFB, Diverted-to user is not subscribed to the		
	COLR service.		
	Ensure that when user A calls user B, the call is forwarded on busy user to user		
	C, if a Redirection number restriction parameter is present it is set to		
	'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE		
	if ISUP/BICC- SIP-I interworking is applicable in Network A.		
Configuration	Subscription options:		
	 Connected user subscribed to COLR = no 		
SIP Parameter	200 OK		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	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)		
	ACK (Call-ID A-B) →		
0	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.		
	Ineheat this test in reverse direction.		

Test case group SIP-SIP/Service/CFB Reference 7.1/ [24] SELECTION EXPRESSION Test purpose SIP-I support. CFB performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'. The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded on busy user to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC - SIP-I intervorking is applicable in Network B. Subscription options: Served user releases his/her number to diverted-to user = Release diverting number information if ISUP/BICC - SIP-I intervorking is applicable in Network B. Subscription options: Served user releases his/her number to diverted-to user = Release diverting number information in the Redirection information if ISUP/BICC - SIP-I intervorking is applicable in Network B. Subscription options: Served user releases his/her number to diverted-to user = Release diverting number information in the Redirection in	Took assa mumban	00 -tl- 040	
Reference 7.1/[24] SELECTION EXPRESSION Network B] SE 17 AND SE 47 AND SE 55 SIP-I support. CFB performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed' The user A and user C are in Network A, The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Served user releases his/her number to diverted-to user a Release diverting number information. Ensure that when user A calls user B, the calls is to warded 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 enapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if SUP/PISCC. SIP-I Intervoking is applicable in Network B. Subscription options:	Test case number	SS_cfb_016	
SELECTION EXPRESSION INterwork B] SE 17 AND SE 55			
SIP-1 support. CFB performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'. The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded on busy user to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC - SIP-1 intervokring is applicable in Network B. Subscription options: Subscription options: Subscription options: Sorved user releases his/her number to diverted-to user = Release diverting number information INVITE Content-Type: multipart/mixed/boundary=[any boundary name] -[any boundary name] Content-Type: multipart/mixed/boundary=[any boundary name] -[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Dispositions isgrial;handling-required IAM Redirecting number Address presentation restricted indicator presentation allowed Address spresentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Redirection information Original redirection information Redirection counter Redirection counter Redirection counter Redirection counter Redirection information Figure 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 address presentation restricted indicator is set to 'presentation allowed'? Check:	Reference		
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. Subscription options: Subscription options: Sorved user releases his/her number to diverted-to user = Release diverting number information 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	SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55	
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 intervorking is applicable in Network B. Subscription options: ■ Served user releases his/her number to diverted-to user = Release diverting number information 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 presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address presentation restricted indicator presentation information Original Redirection information Original Redirection information Original Redirection information Original Redirection reason User Busy [any boundary name]- Message flow SIP (Network A) Interconnection Interface INVITE (Call-ID A-B) CFB is performed INVITE (Call-ID A-B) CFB is performed Originating user in Network A action interface? Check: Is a INVITE request received at the interconnection interface? Check: Is a INVITE request received at the interconnection interface? Check: Is a INVITE request received at the interconnection interface? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Redirection restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Redirecting reason is set to 'User Busy's Check. Is the Circulation and the Redirecting reason	Test purpose	SIP-I support. CFB performed in Network B, Notification of diverted-to user	
Subscription options: Served user releases his/her number to diverted-to user = Release diverting number information 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 signal Redirection information Original Redirection Reason unknown Redirection information Original Redirection counter Redirecting reason User Busy [any boundary name] Message flow SIP (Network A) Interconnection Interface 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 a INVITE request received at the interconnection interface? Check: Is a INVITE request received at the interconnection interface? Check: Is a INVITE request received at the interconnection interface? 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: and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present: and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection information present and the Redirecting reason is set to 'User Busy'?		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	
SiP Parameter Served user releases his/her number to diverted-to user = Release diverting number information Novite	Configuration		
Content-Type: multipart/mixed;boundary=[any boundary name][any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM	Configuration	• Served user releases his/her number to diverted-to user = Release diverting	
Content-Type: multipart/mixed;boundary=[any boundary name][any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM	SIP Parameter	INVITE	
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 Redirecting reason User Busy [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in 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 Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Redirecting reason is set to 'User Busy'?		[any boundary name] Content-Type: application/isup;version=itu-t92	
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 Redirecting reason User Busy [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in 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 Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Redirecting reason is set to 'User Busy'?		IAM	
Address signal (Diverting user) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason User Busy [any boundary name] Message flow SIP (Network A) Interconnection Interface 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 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 and the Redirecting reason is set to 'User Busy'? Check: Is Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to 'User Busy'?		Redirecting number	
Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason User Busy[any boundary name] Message flow SIP (Network A) Interconnection Interface 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 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 Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is Redirection number present? Check: Is Redirection number present and the Redirecting reason is set to 'User Busy'?		presentation allowed	
Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason User Busy[any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in 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? Check: Is Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to 'User Busy'?		Address signal (<i>Diverting user</i>)	
presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason User Busy[any boundary name] Message flow SIP (Network A) Interconnection Interface 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 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 Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Redirecting reason is set to 'User Busy'? Check: Is Redirection information present and the Redirecting reason is set to 'User Busy'?			
Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting indicator Redirecting reason User Busy[any boundary name] Message flow SIP (Network A) Interconnection Interface 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 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 'User Busy'?			
Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason User Busy [any boundary name] Message flow SIP (Network A) Interconnection Interface 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 Redirection 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'?			
Original Redirection Reason unknown Redirecting indicator Redirecting reason User Busy [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A Check: Is a INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is the Redirection information present and the Redirecting reason is set to 'User Busy'?		Address signal	
Redirecting indicator Redirecting reason User Busy [any boundary name] Message flow SIP (Network A) Interconnection Interface 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 an 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? Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to 'User Busy'?		Original Redirection Reason	
Redirection counter Redirecting reason User Busy [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A Check: Is a INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to 'User Busy'?			
Redirecting reason User Busy [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C, IAM) 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: Is a INVITE request received at the interconnection interface? 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? Is Redirection information present and the Redirecting reason is set to 'User Busy'?			
User Busy [any boundary name] Message flow SIP (Network A) Interconnection Interface 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'?			
[any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C, IAM) 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: 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'?			
Message flow SIP (Network A) Interconnection Interface 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'?		User Busy	
Message flow SIP (Network A) Interconnection Interface 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'?			
SIP (Network A) Interconnection Interface 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 the Redirection information present and the Redirecting reason is set to 'User Busy'?		[any boundary name]	
CFB is performed NVITE Call-ID B-C, IAM Apply post test routine	Message flow SIP (Network A)	,	
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'?			
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'?	•	NVITE(Call-ID B-C, IAM)	
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'?			
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'?	Comments		
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'?			
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'?			
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'?			
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'?			
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'?			
Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to 'User Busy'?			
Check: Is Redirection information present and the Redirecting reason is set to 'User Busy'?			
'User Busy'?			
Repeat this test in reverse direction.			
		Repeat this test in reverse direction.	

SS_cfb_017 SS_cfb_017 SIP-SIP/Service/CFB SIP-SIP/Service/CFB SIP-SIP/Service/CFB SELECTION EXPRESSION [Network B] SE 17 AND SE 47 AND SE 55 SIP-I support. CFB performed in Network B, Notification of diverted-transfer Redirecting number 'presentation restricted'. The user A and user C are in Network A. The user B is in the PSTN/PLMN SIP-I support. The user B is in th		
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-the Redirecting number 'presentation restricted'. The user A and user C are in Network A. The user B is in the PSTN/PLMN		
[Network B] SE 17 AND SE 47 AND SE 55 Test purpose SIP-I support. CFB performed in Network B, Notification of diverted-th Redirecting number 'presentation restricted'. The user A and user C are in Network A. The user B is in the PSTN/PLMN		
SIP-I support. CFB performed in Network B, Notification of diverted-t Redirecting number 'presentation restricted'. The user A and user C are in Network A. The user B is in the PSTN/PLMN		
Redirecting number 'presentation restricted'. The user A and user C are in Network A. The user B is in the PSTN/PLMN	o user	
The user A and user C are in Network A. The user B is in the PSTN/PLMN	o user	
	l nart	
of Network B and is provided with CFB, Served user releases his/her num		
diverted-to user = Release diverting number information.		
Ensure that when user A calls user B, the call is forwarded on busy user to	o 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		
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 releases 	ease	
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 Property of the Control of the C		
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 Dedication indicator		
Redirecting indicator Redirection counter		
Redirection counter Redirecting reason		
User Busy		
User Busy		
[any boundary name]		
Message flow		
SIP (Network A) Interconnection Interface SIP (Network	(B)	
INVITE(Call-ID A-B) →	-,	
CFB is performed		
← INVITE(Call-ID B-C, IAM)		
Apply post test routine		
Originating user in Network A establishes a call to user in Network B. Network	work B	
performs the diversion to a user in Network A		
Check: Is a INVITE request received at the interconnection interfac	e ?	
Check: Is an IAM encapsulated in the INVITE?		
Check: Is the Redirecting number present and the Address presentation	n	
restricted indicator is set to 'presentation restricted'?		
Check: Is the Original called number present and the Address presenta	ition	
restricted indicator is set to 'presentation restricted'?		
Check: Is the Redirection number present?		
Check: Is Redirection information present and the Redirecting reason is	s set to	
'User Busy'?		
Repeat this test in reverse direction.		

7.1.5.6.3 Communication Forwarding No Reply (CFNR)

Test case number	SS_cfnr_	001		
Test case group	SIP-SIP/Service/CFNR			
Reference	4.5.2.6/ [9	9]		
SELECTION EXPRESSION	SE 27			
Test purpose	Commur	nication forwarding no reply, ba	sic rules.	
	The user	A and user C are in Network A. T	he user B is	s in network B and is
	1	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		
		Apply post test routine		
Comments	Check:	CDIV no reply is successful.		
	Check:	In the active call state, ensure th		
	Check:	Is the P-Asserted-Identity preser	it set to the	identity of the originating
	Donost th	user?		
	Repeat tr	nis test in reverse direction.		

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

Test case number	SS_cfnr_003		
Test case group	SIP-SIP/Service/CFNR		
Reference	4.5.2.6/ [9]		
SELECTION EXPRESSION	SE 27 AND SE 30		
Test purpose	Communication forwarding no reply, originating user is notified. URI from		
	the served user not received.		
	The user A and user C are in network A. The user B is in network B and is		
	provided with CFNR "Originating user receives notification that his		
	communication has been diverted" = Yes "Served user allows the presentation of		
	forwarded to URI to originating user in diversion notification" = No and. "Served		
	user allows the presentation of his/her URI to originating user in diversion notification" = No.		
	Ensure that when user A calls user B, the call is forwarded no reply to user C,		
	user A is notified of call diversion and not informed of the diverted-to number and		
	served user number.		
Configuration	Subscription options:		
	Originating user receives notification that his communication has been		
	diverted = Yes		
	 Served user allows the presentation of forwarded to URI to originating user 		
	in diversion notification = No		
	 Served user allows the presentation of his/her URI to originating user in 		
	diversion notification = No		
SIP Parameter	181 Being Forwarded		
	<sip:userb@networkb?privacy=history>;index=1,</sip:userb@networkb?privacy=history>		
	<sip: userc@networka;cause="408?Privacy=history">;index=1.1</sip:>		
Message flow	Interconnection Interfere		
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 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.		
	וויפףפגו ווווס נפסג ווו ופייפוסט טוופטנוטוו.		

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:		
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 = Yes 		
SIP Parameter	181 Being Forwarded		
	<pre><sip:userb@networkb>;index=1, <sip: userc@networka;cause="408">;index=1.1</sip:></sip:userb@networkb></pre>		
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 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:	
3	 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< a="">;index=1, <ip:userc@network1;cause=408< a="">;index=1.1</ip:userc@network1;cause=408<></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 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	CC efec 000	
	SS_cfnr_006	
Test case group	SIP-SIP/Service/CFNR	
Reference	4.5.2.6/ [9]	
SELECTION EXPRESSION	SE 27 AND SE 30	
Test purpose	Communication forwarding no reply, diverted-to user receives the URI of the diverted-to user.	
	The user A and user C are in network A. The user B is in network B and is provided with "Served user allows the presentation of his/her URI to the diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded no reply to user C,	
	user C is informed of the forwarding number.	
Configuration	Subscription options:	
	 Served user allows the presentation of his/her URI to the diverted-to user = 	
	Yes	
SIP Parameter	INVITE	
	Request line contains ';cause=408'	
	History-Info header:	
	<sip:userb@networkb>;index=1,</sip: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 stor 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:>				
	1007, 110010 @1101W01W1,000000 = 1007, 11100X = 1.11				
	181 Being Forwarded				
	History-Info header:				
	<pre><sip:userb@network>;index=1,</sip:userb@network></pre>				
	<sip: userc@networka;cause="408">;index=1.1</sip:>				
	solp. door of notwork you door in door in a				
	200 OK INVITE				
	History-Info header:				
	<pre><sip:userb@networkb>;index=1,</sip:userb@networkb></pre>				
	<sip: userc@networka;cause="408">;index=1.1</sip:>				
Message flow	to fire about Streeth and House Took Jindow 111				
SIP (Network A)	Interconnection Interface SIP (Network B)				
J. (1.01.1.7.4)	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-Δ)				
	← 200 OK INVITE(Call-ID B-A)				
	ACK(Call-ID A-B) →				
Comments	ACK(Call-ID A-B) → Apply post test routine				
Comments	ACK(Call-ID A-B) Apply post test routine Check: A History-Info header is received in the INVITE at the interconnection				
Comments	ACK(Call-ID A-B) Apply post test routine 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.				
Comments	ACK(Call-ID A-B) Apply post test routine 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				
Comments	ACK(Call-ID A-B) Apply post test routine 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				
Comments	ACK(Call-ID A-B) Apply post test routine 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.				
Comments	ACK(Call-ID A-B) Apply post test routine 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				
Comments	ACK(Call-ID A-B) Apply post test routine 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'?				
Comments	ACK(Call-ID A-B) Apply post test routine Check: A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. Check: A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. Check: Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '408'? Check: Is the cause parameter in the last entry is set to '408'?				
Comments	ACK(Call-ID A-B) Apply post test routine Check: A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. Check: A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. Check: Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '408'? Check: Is the cause parameter in the last entry is set to '408'? NOTE: The history entries can be accumulated in "one" History-Info header or				
Comments	ACK(Call-ID A-B) Apply post test routine Check: A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. Check: A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. Check: Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '408'? Check: Is the cause parameter in the last entry is set to '408'?				

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

Test case number	SS cfnr	009			
Test case group	SIP-SIP/Service/CFNR				
Reference					
	4.5.2.6/ [9	9]			
SELECTION EXPRESSION	SE 27				
Test purpose	Communication forwarding no reply, unsuccessful NDUB.				
	The user A and user C are in network A. The user B is in network B and is provided with CFNR. Ensure that when user A calls user B, the call is forwarded no reply to user C				
	and user C is network determined user busy.				
Configuration					
SIP Parameter					
Message flow					
SIP (Network A)		Interconnection Interface		SIP (Network B)	
,		INVITE(Call-ID A-B)	→	,	
	←	180 Ringing(Call-ID B-A)			
		CFB is performed			
	←	INVITE(Call-ID B-C)			
		486 Busy Here(Call-ID C-B)	→		
	(ACK(Call-ID B-C)	_		
	(486 Busy Here(Call-ID A-B)			
		ACK(Call-ID A-B)	→		
Comments	Check:	The dialogue is terminated by re	eceiving a 4	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" = No, "diverting number is released to the diverted-to user" = 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 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 potification = No. 			
	 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.			

Toot coop number	CC etax 044				
Test case number	SS_cfnr_011				
Test case group	SIP-SIP/Service/CFNR				
Reference SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55				
Test purpose	SIP-I support. CFNR performed in Network B, Notification subscription options is set to presentation not allowed.				
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number.				
	Ensure that when user A calls user B, the call is forwarded on no reply to user C user A is not notified about call diversion. The notification information is present in the encapsulated CPG contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC				
	interworking is applicable in Network B.				
Configuration	Subscription options:				
	 Calling user receives notification that his call has been diverted (forwarded or deflected) = no 				
SIP Parameter	183 Session Progress				
	Content-Type: multipart/mixed;boundary=[any boundary name]				
	[any boundary name]				
	Content-Type: application/isup;version=itu-t92				
	Content-Type: application/isap,version=ita-ts2 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]				
Message flow					
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing (Call-ID B-A, ACM)				
	CFNR is performed				
	← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, CPG)				
	183 Session Progress (Call-ID B-A, CPG) Apply post test routine				
Comments	Originating user in Network A establishes a call to user in Network B. Network B				
- Comments	performs the diversion to a user in Network A				
	Check: Is a 183 Session Progress received at the interconnection interface?				
	Check: Is an CPG encapsulated in the 183?				
	Check: Is the Called party's status indicator set to 'no indication'?				
	Check: Is the Redirection number present?				
	Check: Is Notification subscription options indicator set to 'presentation not				
	allowed'?				
	Check: Is the Redirecting reason set to 'No reply'?				
	Repeat this test in reverse direction.				

Test case number	SS_cfnr_012			
Test case group	SIP-SIP/Service/CFNR			
Reference	6.5/ [24]			
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55			
Test purpose	SIP-I support. CFNR performed in Network B, Notification subscription options is set to presentation allowed without redirection number.			
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on no reply to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated CPG contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.			
Configuration	Subscription options:			
_	 Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number 			
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required			
	Event indicator Alerting or Progress Redirection number Address signal (Diverted-to user) Call diversion information Notification subscription options presentation allowed without redirection number Redirecting reason No reply Generic notification call is diverting			
Message flow SIP (Network A)	[any boundary name] Interconnection Interface SIP (Network B)			
← ←	INVITE(Call-ID A-B) 180 Ringing (Call-ID B-A, ACM) CFNR is performed INVITE(Call-ID B-C, IAM) 183 Session Progress (Call-ID B-A, CPG) Apply post test routine			
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A Check: 183 Session Progress is received at the interconnection interface. Check: Is an CPG encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator is set to 'presentation allowed without redirection number'? Check: Is the Redirecting reason set to 'No reply'? Repeat this test in reverse direction.			

Test case number	SS_cfnr_013				
Test case group	SIP-SIP/Service/CFNR				
Reference	6.5/ [24]				
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55				
Test purpose	SIP-I support. CFNR performed in Network B, Notification subscription options is set to presentation allowed with redirection number.				
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number. Ensure that when user A calls user B, the call is forwarded on no reply to user C user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated CPG contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.				
Configuration	Subscription options:				
	 Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number 				
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name]				
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required				
	CPG Event indicator Alerting or Progress Redirection number Address signal (<i>Diverted-to user</i>) Call diversion information Notification subscription options				
	presentation allowed with redirection number 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) → ** ** ** ** ** ** ** ** **				
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A Check: 183 Session Progress is received at the interconnection interface. Check: Is an CPG encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator is set to 'presentation allowed with redirection number'? Check: Is the Redirecting reason set to 'No reply'? Repeat this test in reverse direction.				

Test case number	SS_cfnr_014			
Test case group	SIP-SIP/Service/CFNR			
Reference	6.7/ [24]			
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53			
Test purpose	SIP-I support. CFNR performed in Network B, Restriction of the Redirection			
, , , , , , , , , , , , , , , , , , ,	number.			
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part			
	of Network B and is provided with CFNR, Diverted-to user is subscribed to the			
	COLR service in Permanent mode.			
	Ensure that when user A calls user B, the call is forwarded on no reply to user C,			
	a Redirection number restriction parameter is present set to 'Presentation			
	restricted in the encapsulated ANM contained in the 200 OK INVITE if			
	ISUP/BICC- SIP-I interworking is applicable in Network A.			
Configuration	Subscription options:			
	Connected user subscribed to COLR, Permanent = yes			
SIP Parameter	200 OK			
	Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any baymalan mana]			
	[any boundary name]			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	ANM			
	Redirection number restriction			
	Presentation restricted			
	1 100011011010			
	[any boundary name]			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE(Call-ID A-B), IAM →			
	← 180 Ringing (Call-ID B-A, ACM)			
	CFNR is performed			
	← INVITE(Call-ID B-C)			
	180 Ringing (Call-ID C-B, ACM) →			
	180 Ringing (Call-ID B-A)			
	200 OK INVITE (Call-ID C-B, ANM) →			
	← ACK (Call-ID B-C) ← 200 OK INVITE (Call-ID B-A)			
	- 200 011 1111 (Can 12 2 11)			
	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 set to 'Presentation			
	restricted'?			
	Repeat this test in reverse direction.			
	Include this test in reverse direction.			

Test case number	SS_cfnr_015				
Test case group	SIP-SIP/Service/CFNR				
Reference	6.7/ [24]				
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53				
Test purpose	SIP-I support. CFNR performed in Network B, No restriction of the				
rest purpose	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				
0	ISUP/BICC- SIP-I interworking is applicable in Network A.				
Configuration	Subscription options:				
	 Connected user subscribed to COLR = no 				
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name]				
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required				
	ANM				
	Redirection number restriction				
	Presentation allowed				
	or				
	Redirection number restriction not present				
	[any boundary name]				
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B), IAM →				
•	F 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) →				
0	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	CC ofor 016			
Test case group	SS_cfnr_016			
Reference	SIP-SIP/Service/CFNR			
SELECTION EXPRESSION	7.1/ [24] [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 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:			
garanen.	 Served user releases his/her number to diverted-to user = Release diverting number information 			
CID Danamatan				
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			
	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 Redirection information present and the Redirecting reason is set to			
	'No reply'?			
	Repeat this test in reverse direction.			

Test case number	SS_cfnr_017			
Test case group	SIP-SIP/Service/CFNR			
Reference	7.1/ [24]			
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55			
Test purpose	SIP-I support. CFNR performed in Network B, Notification of diverted-to			
	user Redirecting number 'presentation restricted'.			
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part			
	of Network B and is provided with CFNR, Served user releases his/her number			
	to diverted-to user = Release diverting number information.			
	Ensure that when user A calls user B, the call is forwarded on no reply to user C,			
	user C is notified of call diversion and informed of the diverting number.			
	The notification information is present in the encapsulated IAM contained in the			
	Redirecting number 'presentation restricted' and Redirection information if			
	ISUP/BICC - SIP-I interworking is applicable in Network B.			
Configuration	Subscription options:			
	Served user releases his/her number to diverted-to user = Do not release			
OID Developed	diverting numberinformation			
SIP Parameter	INVITE			
	Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name]			
	Content-Type: application/isup;version=itu-t92			
	Content-Type: application/isdp,version=itd-is2			
	Oomon Disposition: Signat, 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			
	Redirecting indicator Redirection counter			
	Redirecting reason No reply			
	No reply			
	[any boundary name]			
Message flow	[
SIP (Network A)	Interconnection Interface SIP (Network B)			
, ,	INVITE(Call-ID A-B) →			
	← 180 Ringing (Call-ID B-A, ACM)			
	CFNR is performed			
	← INVITE(Call-ID B-C, IAM)			
	Apply post test routine			
Comments	Originating user in Network A establishes a call to user in Network B. Network B			
	performs the diversion to a user in Network A			
	Check: Is a INVITE request received at the interconnection interface?			
	Check: Is an IAM encapsulated in the INVITE?			
	Check: Is the Redirecting number present and the Address presentation			
	restricted indicator is set to 'presentation restricted'? Check: Is the Original called number present and the Address presentation			
	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.			
	reposit the test in reverse direction.			

7.1.5.6.4 Communication Forwarding Not Logged in (CFNL)

-	T==				
Test case number		SS_cfnl_001			
Test case group	SIP-SIP/Service/CFNL				
Reference		4.5.2.6/ [9]			
SELECTION EXPRESSION	SE 28	SE 28			
Test purpose	Communication forwarding not logged in, basic rules.				
		A and user C are in Network A. T with CFNL.	he user B i	s in network B and is	
	p	hat when user A calls user B, the o	call is forwa	rded not logged in to user	
		active call state, ensure the prope			
Configuration	00	don'to can ciato, cricaro irio propo	nty or opoo	o	
SIP Parameter					
Message flow					
SIP (Network A)		Interconnection Interface		SIP (Network B)	
,		INVITE(Call-ID A-B)	→	,	
		CFNL is performed			
	←	INVITE(Call-ID B-C)			
		180 Ringing(Call-ID C-B) →			
	←	180 Ringing(Call-ID B-A)			
		200 OK INVITE(Call-ID C-B) →			
	←	ACK(Call-ÌD B-C)			
	←	200 OK INVITE(Call-ID B-A)			
		ACK(Call-ID A-B)	→		
		Communication			
		Apply post test routine			
Comments	Check:	The CDIV not logged in is succes			
	Check:	In the active call state, ensure the			
	Check:	Is the P-Asserted-Identity present set to the identity of the originating			
		user?			
	Repeat th	his test in reverse direction.			

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

Test case number	SS_cfnl_003
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/ [9]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	Communication forwarding not logged in, originating user is notified. URI
l con pui poss	of the diverted-to user not received.
	The user A and user C are in network A. The user B is in network B and is provided with CFNL "Originating user receives notification that his communication has been diverted" = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and not informed of the diverted-to number and the served user number.
Configuration	Subscription options:
	 Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to originating user in diversion notification = No
SIP Parameter	181 Being Forwarded
on rarameter	<sip:userb@networkb?<mark>Privacy=history>;index=1,</sip:userb@networkb?<mark>
	<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 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 '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:
Conniguration	 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:></sip:userb@networkb>
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)
	4 180 Ringing(Call-ID B-A)
Comments	Apply post test routine 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:
OID D	• Served user allows the presentation of his/her URI to diverted-to user = No
SIP Parameter	INVITE
	Request line contains ';cause=404'
	History-Info header: <sip:userb@networkb?privacy=history>;index=1,</sip:userb@networkb?privacy=history>
	<sip: userc@network1;cause="404">;index=1.1</sip:>
Message flow	T CSIp. doe! 0 @ network 1, cadoe=4042, index=1.1
SIP (Network A)	Interconnection Interface SIP (Network B)
Cii (Hotillorik 71)	INVITE(Call-ID A-B) →
	CFNL is performed
	← INVITE(Call-ID B-C)
	Apply post test routine
Comments	Check: A History-Info header is received in the INVITE contains the URI of
	user B (served user) at the interconnection interface and a Privacy
	header is escaped set to 'history'.
	Check: Is the 'cause' parameter present in the Request line sent to user C
	(diverted-to user) set to '404'?
	Check: Is the cause parameter in the last entry is set to '404'?
	NOTE: The history entries can be accumulated in "one" History-Info header or
	each history entry is present in one single History-Info header. Repeat this test in reverse direction.
	Nepeat this test in reverse direction.

Test case number	SS_cfnl_006
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/ [9]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	Communication forwarding not logged in, diverted-to user receives the URI
	of the served user.
	The user A and user C are in network A. The user B is in network B and is
	provided with CFNL "Served user allows the presentation of his/her URI to
	diverted-to user" = Yes.
	Ensure that when user A calls user B, the call is forwarded not logged in to user
	C, user C is informed of the forwarding number.
Configuration	Subscription options:
	 Served user allows the presentation of his/her URI to diverted-to user = Yes
SIP Parameter	INVITE
	Request line contains ';cause=404'
	History-Info header:
	<sip:userb@networkb>,index=1,</sip:userb@networkb>
	<sip: userc@networka;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)
	← 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:
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 = Yes diverting number is released to the diverted-to user = Yes
SIP Parameter	INVITE:
	Request line contains ';cause=404' History-Info header: < sip:userB@networkB&Reason=SIP;cause=404 ;index=1, < sip:userC@networkA;cause=404 ;index=1.1
	181 Being Forwarded
	History-Info header: <sip:userb@network>;index=1, <sip: userc@networka;cause="404">;index=1.1 200 OK INVITE History-Info header: <sip:userb@networkb>;index=1, <sip: userc@networka;cause="404">;index=1.1</sip:></sip:userb@networkb></sip:></sip:userb@network>
Message flow	The state of the s
SIP (Network A)	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) COUNTY INVITE(Call-ID B-A) ACK(Call-ID A-B) Apply post test routine
Comments	Check: A History-Info header is received in the INVITE at the interconnection
	 interface sent to user C containing the URI identifying the served user. Check: A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. Check: Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '404'? Check: Is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or
	each history entry is present in one single History-Info header.
	Repeat this test in reverse direction.

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

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

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

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

Test case number	SS_cfnl_012					
Test case group	SIP-SIP/Service/CFNL					
Reference	6.5/ [24]					
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55					
Test purpose	SIP-I support. CFNL performed in Network B, Notification subscription options is set to presentation allowed without redirection number.					
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the					
	Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.					
Configuration	Subscription options:					
	 Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number 					
SIP Parameter	183 Session Progress					
	Content-Type: multipart/mixed;boundary=[any boundary name]					
	[any boundary name]					
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required					
	Content-Disposition: signal;nandling=required					
	ACM					
	Backward call indicator Called party's status indicator no indication Redirection number					
	Address signal (Diverted-to user)					
	Call diversion information					
	Notification subscription options presentation allowed without redirection number Redirecting reason					
	Mobile subscriber not reachable Generic notification call is diverting[any boundary name]					
Message flow	[arry boundary mame]					
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 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.					
	The Part and test in test seasons and seasons					

Tool coop number	CC atal 042				
Test case number	SS_cfnl_013				
Test case group	SIP-SIP/Service/CFNL				
Reference	6.5/ [24]				
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55				
Test purpose	SIP-I support. CFNL performed in Network B, Notification subscription options is set to presentation allowed with redirection number.				
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number.				
	Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC				
	interworking is applicable in Network B.				
Configuration	Subscription options:				
	• Calling user receives notification that his call has been diverted (forwarded				
CID Devemeter	or deflected) = yes, with diverted-to user number				
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name][any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required				
	ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (<i>Diverted-to user</i>) Call diversion information Notification subscription options				
	presentation allowed with redirection number Redirecting reason Mobile subscriber not reachable Generic notification call is diverting				
Message flow	[any boundary name]				
SĪP (Network A)	Interconnection Interface 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 routing				
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: 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:						
010.0	 Connected user subscribed to COLR, Permanent = yes 						
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name]						
	[any boundary name] Content-Type: application/isup;version=itu-t92						
	Content-Type: application/isup,version=itu-tis2 Content-Disposition: signal;handling=required						
	Content-Disposition: signal;nandling=required						
	ANM						
	Redirection number restriction						
	Presentation restricted						
	1 1000 Matter 100th Cod						
	[any boundary name]						
Message flow							
SIP (Network A)	Interconnection Interface SIP (Network B)						
	INVITE(Call-ID A-B), IAM →						
	CFNL is performed						
•	► INVITE(Call-ID B-C)						
	180 Ringing (Call-ID C-B, ACM) →						
•	+ 180 Ringing (Call-ID B-A)						
	200 OK INVITE (Call-ID C-B, ANM) →						
	ACK (Call-ID B-C)						
·	← 200 OK INVITE (Call-ID B-A)						
	ACK (Call-ID A-B) →						
	Apply post test routine						
Comments	Originating user in Network A establishes a call to user in Network B. Network B						
	performs the diversion to a user in Network A						
	Check: Is a 200 OK INVITE received at the interconnection interface						
	Check: Is an ANM encapsulated in the 200 OK? Check: Is the ISUR/PICC Redirection number restriction act to Presentation						
	Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation						
	restricted'?						
	Repeat this test in reverse direction.						

Test case number	SS_cfnl_015					
Test case group	SIP-SIP/Service/CFNL					
Reference	6.7/ [24]					
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53					
Test purpose	SIP-I support. CFNL performed in Network B, No restriction of the Redirection number.					
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Diverted-to user is not subscribed to the COLR service.					
	Ensure that when user A calls user B, the call is forwarded not logged in to user C, if a Redirection number restriction parameter is present it is set to 'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.					
Configuration	Subscription options:					
garanon	 Connected user subscribed to COLR = no 					
SIP Parameter	Contracted user subscribed to COLR = no 200 OK Content-Type: multipart/mixed;boundary=[any boundary name]					
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required					
	ANM Redirection number restriction Presentation allowed or Redirection number restriction not present					
	[any boundary name]					
	Interconnection Interface 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:				
	• 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-Type: application/isup,version=td-t32 Content-Disposition: signal;handling=required				
	TAXA				
	IAM Dedinaction runshes				
	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					
SIP (Network A)	Interconnection Interface SIP (Network B)				
	INVITE(Call-ID A-B) →				
	CFNL is performed				
	← INVITE(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	[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 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:				
John Marianon	• 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]				
	[any boundary name] Content-Type: application/isup;version=itu-t92				
	Content-Type: application/isup, version=tu-te2 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				
	Mobile subscriber not reachable				
	[any boundary name]				
Message flow					
SIP (Network A)	Interconnection Interface SIP (Network B)				
	INVITE(Call-ID A-B) →				
	CFNL is performed				
•	(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.				
	I. referre and recent returns an assisting				

7.1.5.6.5 Communication Deflection

Test case group Reference 4.5.2.6/ [9] SELECTION EXPRESSION SE 29 Test purpose Communication deflection during alerting, basic rules. The user A and user C are in Network A. The user B is in network B and is provided with CDa. Ensure that when user A calls user B, the call is deflected during alerting to user C. In the active call state, ensure the property of speech. Configuration SIP Parameter Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed	Test case number	SS cd 0	01					
Reference 4.5.2.6/ [9] SELECTION EXPRESSION SE 29 Test purpose Communication deflection during alerting, basic rules. The user A and user C are in Network A. The user B is in network B and is provided with CDa. Ensure that when user A calls user B, the call is deflected during alerting to user C. In the active call state, ensure the property of speech. Configuration SIP Parameter Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed								
SELECTION EXPRESSION SE 29 Test purpose Communication deflection during alerting, basic rules. The user A and user C are in Network A. The user B is in network B and is provided with CDa. Ensure that when user A calls user B, the call is deflected during alerting to user C. In the active call state, ensure the property of speech. Configuration SIP Parameter Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed								
Test purpose Communication deflection during alerting, basic rules. The user A and user C are in Network A. The user B is in network B and is provided with CDa. Ensure that when user A calls user B, the call is deflected during alerting to user C. In the active call state, ensure the property of speech. Configuration SIP Parameter Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed			ชา					
The user A and user C are in Network A. The user B is in network B and is provided with CDa. Ensure that when user A calls user B, the call is deflected during alerting to user C. In the active call state, ensure the property of speech. Configuration SIP Parameter Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed								
provided with CDa. Ensure that when user A calls user B, the call is deflected during alerting to user C. In the active call state, ensure the property of speech. Configuration SIP Parameter Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed	Test purpose	Commur	nication deflection during alertir	ıg, basic rı	ules.			
provided with CDa. Ensure that when user A calls user B, the call is deflected during alerting to user C. In the active call state, ensure the property of speech. Configuration SIP Parameter Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed								
Ensure that when user A calls user B, the call is deflected during alerting to user C. In the active call state, ensure the property of speech. Configuration SIP Parameter Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed								
user C. In the active call state, ensure the property of speech. Configuration SIP Parameter Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed		p						
Configuration SIP Parameter Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed SIP (Network B)								
SIP Parameter Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed		user C. In the active call state, ensure the property of speech.						
Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CDa is performed SIP (Network B) CDa is performed	Configuration							
SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CDa is performed	SIP Parameter							
INVITE(Call-ID A-B) → CDa is performed	Message flow							
CDa is performed	SIP (Network A)		Interconnection Interface		SIP (Network B)			
			INVITE(Call-ID A-B)	→				
← 180 Ringing(Call-ID R-A)			CDa is performed					
		(180 Ringing(Call-ID B-A)					
← INVITE(Call-ID B-C)		(INVITE(Call-ID B-C)					
180 Ringing(Call-ID C-B) →			180 Ringing(Call-ID C-B)	→				
← 180 Ringing(Call-ID B-A)		←	180 Ringing(Call-ID B-A)					
200 OK IŇVIŤĖ(Call-ID C-B) →				→				
← ACK(Call-ID B-C)		←	ACK(Call-ID B-C)					
← 200 OK INVITE(Call-ID B-A)		←	200 OK INVITE(Call-ID B-A)					
ACK(Call-ÌD A-B) →				→				
Communication			Communication					
Apply post test routine			Apply post test routine					
Comments Check: CDa is successful.	Comments	Check:	CDa is successful.					
Check: In the active call state, ensure the property of speech.		Check:	In the active call state, ensure the	e property of	of speech.			
Check: Is the P-Asserted-Identity present set to the identity of the originating		Check:	Is the P-Asserted-Identity presen	it set to the	identity of the originating			
user?			user?		-			
Repeat this test in reverse direction.		Repeat th	nis test in reverse direction.					

Test case number	SS cd 0	02				
Test case group		Service/CD				
Reference						
	4.5.2.6/ [9	9]				
SELECTION EXPRESSION	SE 29					
Test purpose	Commun	ication deflection immediate, b	oasic rules.	•		
	The user A and user C are located in Network A. The user B is located in network B and is provided with CDi. Ensure that when user A calls user B which deflects immediately the communication towards user C (i.e. before alerting starts), the call is forwarded to user C. In the active call state, ensure the property of speech.					
Configuration						
SIP Parameter						
Message flow SIP (Network A)	+ + +	Interconnection Interface INVITE(Call-ID A-B) CDi is performed INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) ACK(Call-ID B-C) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) Communication Apply post test routine	→ → →	SIP (Network B)		
Comments	Check:	heck: CDi is successful. heck: In the active call state, ensure the property of speech.				

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 located in Network A. The user B is located in network B and is provided with CDi, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B which deflects immediately the communication towards user C (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		
1 3 3 4 5 3 1	notified. URI of the diverted-to user not received.		
	The user A and user C are located in network A. The user B is located in		
	network B and is provided with CDi "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 immediately the		
	communication towards user C (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.)	<pre><sip: userc@networka;cause="480?Privacy=history">;index=1.1</sip:></pre>		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)		
SIP (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 A. The user B is in network B and is provided with CDi "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 immediately the communication towards user C (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.		
	וויסף במני ווויס נפסני וווי ובייבוסב עוובטנוטוו.		

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		
l est pui pose	receive the URI of the served user.		
	leading the ord or the served aser.		
	The user A and user C are in network A. The user B is in network B and is		
	provided with CDi "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 immediately the		
	communication towards user C (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 CDi "Served user allows the presentation of his/her URI to diverted-to user" = Yes.		
	Ensure that when user A calls user B which deflects immediately the communication towards user C (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 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_00	8		
Test case group	SIP-SIP/S	ervice/CD		
Reference	4.5.2.6/ [9]		
SELECTION EXPRESSION	SE 29			
Test purpose	Communication Deflection immediate response, unsuccessful UDUB.			
	The user A	A and user C are in network A. T	he user B is ir	n network B and is
	provided v	vith CDi.		
	Ensure that	at when user A calls user B, the	call is deflecte	d immediate to user C
	user C is u	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 re	eceiving a 486	Busy Here.
	Repeat thi	s test in reverse direction.		

SS_cd_009			
SIP-SIP/Service/CD			
4.5.2.6/ [4.5.2.6/ [9]		
SE 29	-		
Commu	Communication Deflection immediate response, unsuccessful NDUB.		
		•	
The user	A and user C are in network A. T	he user B is	s in network B.
Ensure tl	hat when user A calls user B, the	call is deflec	cted immediate to user C
and user	C is network determined user bus	sy.	
		-	
	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		
Check:	The dialogue is terminated by re	eceiving a 4	86 Busy Here.
Repeat t	his test in reverse direction		
	SIP-SIP/ 4.5.2.6/ [SE 29 Commun The user Ensure thand user	SIP-SIP/Service/CD 4.5.2.6/ [9] SE 29 Communication Deflection immediate read to the user A and user C are in network A. The sum of the user A calls user B, the cand user C is network determined user bused and user C is network determined user bused and user C is network determined user bused in the sum of the sum	SIP-SIP/Service/CD 4.5.2.6/ [9] SE 29 Communication Deflection immediate response, u The user A and user C are in network A. The user B is Ensure that when user A calls user B, the call is deflet and user C is network determined user busy. Interconnection Interface 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 Check: The dialogue is terminated by receiving a 4

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 CDi "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, "diverting number is released to the diverted-to user" = No. 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 INVITE(Call-ID A-B) CDi 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. Repeat this test in reverse direction.		

Test case number	SS_cd_011			
Test case group	SIP-SIP/Service/CD			
Reference	6.5/ [24]			
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55			
Test purpose	SIP-I support. CD performed in Network B, Notification subscription			
rest purpose	options is set to presentation not allowed.			
	opilono lo dal la procentation not all'ondal			
	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 or 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 /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			
	Redirection number Address signal (<i>Diverted-to user</i>)			
	Call diversion information			
	Notification subscription options			
	presentation not allowed			
	Redirecting reason			
	Deflection immediate or Deflection during alerting			
	Generic notification			
	call is diverting			
	[any boundary name]			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE(Call-ID A-B)			
	 180 Ringing (Call-ID B-A, ACM) in case CDa CD is performed 			
	← INVITE(Call-ID B-C, IAM)			
	183 / 180 (Call-ID B-A, ACM/CPG)			
	Apply post test routine			
Comments	Originating user in Network A establishes a call to user in Network B. Network B			
	performs the diversion to a user in Network A			
	Check: Is a 183 Session Progress received at the interconnection interface?			
	Check: Is an ACM encapsulated in the 183?			
	Check: Is the Called party's status indicator set to 'no indication'?			
	Check: Is the Redirection number present?			
	Check: Is Notification subscription options indicator set to 'presentation not			
	allowed'?			
	Check: Is the Redirecting reason set to 'Deflection immediate' or 'Deflection			
	during alerting'?			
	Repeat this test in reverse direction.			

Test case number	SS cd 012		
Test case group	SIP-SIP/Service/CD		
Reference	6.5/ [24]		
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55		
Test purpose	SIP-I support. CD performed in Network B, Notification subscription options is set to presentation allowed without redirection number.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number.		
	Ensure that when user A calls user B, the call is deflected to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM or CPG contained in the Redirection number and Call diversion information if SIP-I -		
	ISUP/BICC interworking is applicable in Network B.		
Configuration	Subscription options:		
	Calling user receives notification that his call has been diverted (forwarded)		
	or deflected) = yes, without diverted-to user number		
SIP Parameter	183 /180 Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	ACM/CPG		
	Redirection number		
	Address signal (Diverted-to user)		
	Call diversion information		
	Notification subscription options		
	presentation allowed without redirection number		
	Redirecting reason		
	Deflection immediate or Deflection during alerting		
	Generic notification		
	call is diverting		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →		
	← 180 Ringing (Call-ID B-A) in case CDa CD is performed		
	← INVITE(Call-ID B-C, IAM)		
	← 183 / 180 (Call-ID B-A, ACM/CPG)		
	Apply post test routine		
Comments	Originating user in Network A establishes a call to user in Network B. Network B		
	performs the diversion to a user in Network A		
	Check: 183 Session Progress is received at the interconnection interface.		
	Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'?		
	Check: Is the 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.		
	proposition tool in rotatoo 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 con lo procentation anomou man roun oculon number.			
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part			
	of Network B and is provided with CDi or CDa, Calling user receives notification			
	that his call has been diverted (forwarded or deflected) = yes, with diverted-to			
	user number.			
	Ensure that when user A calls user B, the call is deflected to user C, user A is			
	notified of call diversion and informed of the diverted-to number.			
	The notification information is present in the encapsulated ACM or CPG			
	contained in the Redirection number and Call diversion information if SIP-I -			
	ISUP/BICC interworking is applicable in Network B.			
Configuration	Subscription options:			
	 Calling user receives notification that his call has been diverted (forwarded 			
	or deflected) = yes, with diverted-to user number			
SIP Parameter	183 /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 (<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			
	Call is diverting			
	[any boundary name]			
Message flow	[any boundary name]			
SIP (Network A)	Interconnection Interface SIP (Network B)			
,	INVITE(Call-ID A-B) →			
	← 180 Ringing (Call-ID B-A) in case CDa			
	ČD is performed			
	← INVITE(Call-ID B-C, IAM)			
	← 183 / 180 (Call-ID B-A, ACM/CPG)			
	Apply post test routine			
Comments	Originating user in Network A establishes a call to user in Network B. Network B			
	performs the diversion to a user in Network A			
	Check: 183 Session Progress is received at the interconnection interface.			
	Check: Is an ACM encapsulated in the 183?			
	Check: Is the Called party's status indicator set to 'no indication'?			
	Check: Is the Redirection number present?			
	Check: Is Notification subscription options indicator is set to 'presentation			
	allowed with redirection number'?			
	Check: Is the Redirecting reason set to 'Deflection immediate' or 'Deflection			
	during alerting'?			
	Repeat this test in reverse direction.			

Test case number	SS_cd_014		
Test case group	SIP-SIP/Service/CD		
Reference	6.7/ [24]		
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53		
Test purpose	SIP-I support. CD performed in Network B, Restriction of the Redirection		
	number.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part		
	of Network B and is provided with CDi or CDa, Diverted-to user is subscribed to		
	the COLR service in Permanent mode.		
	Ensure that when user A calls user B, the call is deflected to user C, a Redirection number restriction parameter is present set to 'Presentation		
	restricted' in the encapsulated ANM contained in the 200 OK INVITE if		
	ISUP/BICC- SIP-I interworking is applicable in Network A.		
Configuration	Subscription options:		
3	 Connected user subscribed to COLR, Permanent = yes 		
SIP Parameter	200 OK		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	ANM		
	Redirection number restriction		
	Presentation restricted		
	[24]		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B), IAM → 180 Ringing (Call-ID B-A) in case CDa		
	← 180 Ringing (Call-ID B-A) in case CDa CD is performed		
	← INVITE(Call-ID B-C)		
	180 Ringing (Call-ID C-B, ACM) →		
	← 180 Ringing (Call-ID B-A)		
	200 OK INVITE (Call-ID C-B, ANM) →		
	ACK (Call-ID B-C)		
	← 200 OK INVITE (Call-ID B-A)		
	ACK (Call-ID A-B) →		
_	Apply post test routine		
Comments	Originating user in Network A establishes a call to user in Network B. Network B		
	performs the diversion to a user in Network A Check: Is a 200 OK INVITE received at the interconnection interface?		
	Check: Is a 200 OK INVITE received at the interconnection interface? Check: Is an ANM encapsulated in the 200 OK?		
	Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation		
	restricted'?		
	Repeat this test in reverse direction.		

Test case number	SS_cd_015
Test case group	SIP-SIP/Service/CD
Reference	6.7/ [24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	SIP-I support. CD performed in Network B, No restriction of the Redirection
Tool purpose	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:
Jonnigaranon	Connected user subscribed to COLR = no
SIP Parameter	200 OK
	Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name]
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ANM
	Redirection number restriction
	Presentation allowed
	or
	Redirection number restriction not present
	[any boundary name]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B), IAM →
	INVITE(Call-ID A-B), IAM → 180 Ringing (Call-ID B-A) in case CDa
	CD is performed
	← INVITE(Call-ID B-C)
	180 Ringing (Call-ID C-B, ACM) →
	180 Ringing (Call-ID B-A)
	200 OK INVITE (Call-ID C-B, ANM) →
	ACK (Call-ID B-C)
	← 200 OK INVITE (Call-IĎ 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	
Test purpose	SIP-I support. CD 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 CDi or CDa, Served user releases his/her
	number to diverted-to user = Release diverting number information.
	Ensure that when user A calls user B, the call is deflected to user C, user C is
	notified of call diversion and informed of the diverting number.
	The notification information is present in the encapsulated IAM contained in the
	Redirecting number 'presentation allowed' and Redirection information if
	ISUP/BICC - SIP-I interworking is applicable in Network B.
Configuration	Subscription options:
	• Served user releases his/her number to diverted-to user = Release diverting
	number information
SIP Parameter	INVITE
	Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name]
	Content-Type: application/isup;version=itu-t92
	Content-Type: application/isdp,version=itd-ts2
	Someth Biopositions digital, nationing Froquitou
	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
	Redirecting indicator Redirection counter
	Redirection counter
	Deflection immediate or Deflection during alerting
	2 511 5 511 5 11 5 11 5 11 5 11 5 11 5
	[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
	(Call-ID B-C, IAM)
0	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 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.
	- · ·

Test case number	SS_cd_017
Test case group	SIP-SIP/Service/CD
Reference	7.1/[24]
SELECTION EXPRESSION	
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
	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 (<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
	Redirection counter Redirecting reason
	Deflection immediate or Deflection during alerting
	Bonoolion inimodiate of Bonoolion during diorang
	[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 Redirection information present and the Redirecting reason is set to
	'Deflection immediate' or 'Deflection during alerting'?
	Repeat this test in reverse direction.
	· ·

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

SIP (Network A) Establish a confirmed session to user B1 from Network A to Network B and put it on hold session to user B2 from Network A to Network B and put it on hold User A establishes a 3PTY conversation **REFER** (user B1)	Message flow				
Establish a confirmed session to user B2 from Network A to Network B and put it on hold User A establishes a 3PTY conversation REFER (user B1)	SIP (Network A)		Interconnection Interface		SIP (Network B)
User A establishes a 3PTY conversation REFER (user B1)	Establish a confirmed s	ession	to user B1 from Network A to N	letwork B ar	nd put it on hold
REFER(user B1)	Establish a confirmed s	session	to user B2 from Network A to N	Network B ar	nd put it on hold
## Comments ## Copy Accepted ## NOTIFY(B1, 100)		User A	establishes a 3PTY conversat	ion	
HOTIFY(B1, 100) 200 OK NOTIFY INVITE(focus, user B1) 200 INVITE ACK NOTIFY(B1, 200) 200 OK NOTIFY BYE(user B1) COMMOTIFY BYE(user B1) COMMOTIFY BYE(user B2) INVITE(focus, user B2) ON OK NOTIFY INVITE(focus, user B2) 200 OK NOTIFY INVITE(focus, user B2) 200 INVITE ACK NOTIFY(B2, 200) 200 OK NOTIFY BYE(user B2) 200 INVITE ACK NOTIFY(B2, 200) 200 OK NOTIFY BYE(user B2) COMMOTIFY BYE(user B2) Apply post test routine Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.			REFER(user B1)	→	
## Tool of Notify ## Tool of the confirmed communication to user B1 and B2 contains the URI of the conference focus and is the method parameter to the conference focus and is the method parameter to the conference focus and is the method parameter the conference focus and is the method parameter the conference focus and is the method parameter to the conference focus and is the conference focus and	•	←	202 Accepted		
## INVITE(focus, user B1) 200 INVITE ACK NOTIFY(B1, 200) 200 OK NOTIFY BYE(user B1) 200 OK BYE ### 202 Accepted NOTIFY(100) 200 OK NOTIFY 1NVITE(focus, user B2) 200 INVITE ACK NOTIFY(B2, 200) 200 OK NOTIFY ACK NOTIFY(B2, 200) 200 OK NOTIFY BYE(user B2) 200 INVITE ACK NOTIFY(B2, 200) 200 OK NOTIFY BYE(user B2) 200 OK NOTIFY BYE(user B2) 200 OK BYE Apply post test routine Comments	•	←	NOTIFY(B1, 100)		
## ACK ## NOTIFY(B1, 200) 200 OK NOTIFY BYE(user B1) ## 200 OK BYE ## 202 Accepted NOTIFY(100) 200 OK NOTIFY ** INVITE(focus, user B2) ## ACK NOTIFY(B2, 200) 200 OK NOTIFY ## ACK NOTIFY(B2, 200) 200 OK NOTIFY ## ACK ## NOTIFY(B2, 200) 200 OK NOTIFY BYE(user B2) ## 200 OK BYE ## Apply post test routine Comments			200 OK NOTIFY	→	
Comments ← ACK ← NOTIFY(B1, 200) 200 OK NOTIFY → BYE(user B1) → EVECUTE: 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) → EVECUTE: EVECUTE: APPLY post test routine Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.	•	←	INVITE(focus, user B1)		
## NOTIFY(B1, 200) 200 OK NOTIFY BYE(user B1) ## 200 OK BYE ## 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 NOTIFY ## ACK ## NOTIFY(B2, 200) 200 OK NOTIFY ## BYE(user B2) ## 200 OK BYE ## Apply post test routine ## Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.			200 INVITE	→	
200 OK NOTIFY BYE(user B1) COMMENT REFER (user B2) COMMENT REFER (use	•	←	ACK		
## BYE(user B1)	•	←	NOTIFY(B1, 200)		
REFER (user B2) Comments REFER (user B2) Comments REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter REFER (user B2) Contains the URI of the conference focus and is the method parameter.			200 OK NOTIFY		
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 NOTIFY BYE (user B2) Apply post test routine Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.				→	
Comments ← 202 Accepted ← NOTIFY(100) 200 OK NOTIFY ← INVITE(focus, user B2) 200 INVITE → ACK ← ACK ← NOTIFY(B2, 200) 200 OK NOTIFY → BYE(user B2) → 200 OK BYE Apply post test routine User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.	•	←	200 OK BYE		
Comments ← 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 User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.			REFER(user B2)	4	
 NOTIFY(100) 200 OK NOTIFY → INVITE(focus, user B2)	,	~		•	
200 OK NOTIFY INVITE(focus, user B2) 200 INVITE ACK NOTIFY(B2, 200) 200 OK NOTIFY BYE(user B2) 200 OK BYE Apply post test routine Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.					
## INVITE(focus, user B2) 200 INVITE ACK NOTIFY(B2, 200) 200 OK NOTIFY BYE(user B2) 200 OK BYE Apply post test routine ### Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.		•	` ,	->	
200 INVITE ACK NOTIFY(B2, 200) 200 OK NOTIFY BYE(user B2) Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.		←		•	
ACK NOTIFY(B2, 200) 200 OK NOTIFY BYE(user B2) Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.		•		->	
WOTIFY(B2, 200) 200 OK NOTIFY BYE(user B2) Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.		(-	
200 OK NOTIFY BYE(user B2) 200 OK BYE 200 OK BYE Apply post test routine Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.					
BYE(user B2) 200 OK BYE Apply post test routine Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.			* * *	→	
Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.					
Apply post test routine Comments User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.	•	←			
User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.					
user B1 and B2 are set on hold Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.	Comments	Jser A es		after the confi	rmed communication to
Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter.					
·				FER method s	sent to user B1 and B2
·			contains the URI of the conferer	nce focus and	I is the method parameter
JOHN THE .			set to 'INVITE'.		, , , , ,
Check: The NOTIFY after the REFER request contains the 'SIP/2.0 100'	lc	Check:		equest contai	ns the 'SIP/2.0 100'
message body.				•	
Check: The INVITE request is sent by user B1 and user B2 to the conference	lc	Check:		iser B1 and u	ser B2 to the conference
focus the Request URI is used from the Refer-To header of the					
received REFER request.					
Check: The NOTIFY after the REFER request contains the 'SIP/2.0 200 OK	lc lc	Check:		equest contai	ns the 'SIP/2.0 200 OK'
message body.					
Check: The original session is terminated by user A.	lc lc	Check:		ed by user A.	
Repeat this test in reverse direction.	R	Repeat th	· ·	·	

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.
	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
	 Ensure that user A can invite user B1 to the conference by sending a reINVITE request.
	 Ensure that user A can invite user B2 to the conference by sending a reINVITE request.
Configuration	=
SIP Parameter	INVITE <b1> From: <usera> To: <userb1></userb1></usera></b1>
	Call-ID: A-B1 P-Asserted-Identity: <usera></usera>
	SDP: a=sendrecv
	INVITE <b2> From: <usera> Call-ID: A-B2 To: <userb2> P-Asserted-Identity: <usera></usera></userb2></usera></b2>
	SDP: a=sendrecv
	Interconnection Interface ed session to user B1 from Network A to Network B and put it on hold ed 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)
	INVITE(Call-ID A-B2) → 200 INVITE ACK →
Comments	Apply post test routine User A establishes a 3PTY conversation after the confirmed communication to
Comments	user B1 and B2 are set on hold Check: An INVITE is sent to user B1 and user B2 indicating a new IP address in the 'c' line of the SDP. Check: The 'a' line indicates 'sendrecv'.
	Repeat this test in reverse direction.

Test case number	SS conf 003
Test case group	SIP-SIP/Service/CONF
Reference	
	5.4/ [24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	SIP-I/ISUP interworking. Served user establishes a 3 Party communication.
	Served User A is located in Network A and ISUP/BICC - SIP-I interworking
	applies in Network A. User A establishes a confirmed communication with a User
	B1 in Network B and sets it on HOLD. User A establishes a confirmed
	communication with a User B2 in Network B.
	Ensure that when User A establishes a 3 PTY communication:
	 an INFO request is sent to User B1 in Network B and a ISUP/BICC CPG
	is encapsulated the Generic Notification is set to 'conference
	established';
	an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG
	is encapsulated the Generic Notification is set to 'conference
	established'.
Configuration	ISUP/BICC interworking applies in Network A
Comiguration	User in Network A is subscribed to the 3PTY supplementary service
SIP Parameter	
SIP Parameter	INFO <b1></b1>
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ODO.
	CPG
	Generic Notification
	Conference established
	INFO <b2></b2>
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	200
	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
Establish a conf	firmed session from User A in Network A to user B2 in Network B
	<mark>INFO</mark> (Call-ID A-B1, <mark>CPG</mark>) →
	← 200 INFO
	INFO(Call-ID A-B2, CPG) →
	← 200 INFO
	Apply post test routine
Comments	User A establishes confirmed communication to user B1 in Network B and sets it
	on hold
	User A establishes a confirmed communication to user B2 in Network B
	User A invokes the 3PTY communication
	Check: Is an INFO request sent to user B1 and user B2 in Network B?
	Check: Is a ISUP/BICC CPG message encapsulated in the INFO request to
	both remote users in Network B?
	Check: Is the Generic Notification parameter in the encapsulated CPG in both
	INFO set to 'Conference established'?
	Repeat this test in reverse direction.

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.
	· ·
	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; A BYE request is sent to User B1 in Network B; A BYE request is sent to User B1 in Network B; A BYE request is sent to User B1 in Network B; A BYE request is sent to User B1 in Network B; A BYE request is sent to User B1 in Network B; 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
Comigaration	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
	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
Establish a confil	med session from User A in Network A to user B2 in Network B User A establishes a 3PTY conversation
	BYE(Call-ID A-B1, REL)
	€ 200 INFO
	200 1141 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.
	וויסף במני וווי נפטנ וווי ופייסוסט עווסטנוטוו.

Test case number	SS_conf_005
Test case group	SIP-SIP/Service/CONF
Reference	5.4/ [24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	SIP-I/ISUP interworking. Served user splits the 3 Party communication.
	Served User A is located in Network A and ISUP/BICC - SIP-I interworking applies in Network A. User A establishes a confirmed communication with a User
	B1 in Network B and sets it on HOLD. User A establishes a confirmed
	communication with a User B2 in Network B. User A invokes 3PTY conversation
	Ensure that when User A splits the 3 PTY communication:
	an INFO request is sent to User B1 in Network B and a ISUP/BICC CPG
	is encapsulated the Generic Notification is set to 'Conference disconnected';
	 an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG
	is encapsulated the Generic Notification is set to 'Conference
	disconnected'.
Configuration	ISUP/BICC interworking applies in Network A
- Comigaration	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
	Someth Bioposition: Signati, narrating Froquired
	CPG
	Generic Notification
	Conference disconnected
	Commonded discontinuous
	INTO .P.O.
	INFO <b2> Content-Type: application/isup;version=itu-t92</b2>
	Content-Type: application/isup,version=itu-t92 Content-Disposition: signal;handling=required
	Content-Disposition: signal, nanding-required
	CPG
	Generic Notification
	Conference disconnected
Message flow	
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 irmed session from User A in Network A to user B2 in Network B
Establish a com	User A establishes a 3PTY conversation
	INFO(Call-ID A-B1, CPG) → 200 INFO
	€ 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.
	Inoposit and tost in reverse uncouldn.

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	
3	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 Conserie Netification	
	Generic Notification conference established	
	conference established	
	INFO2 <b1></b1>	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	CPG	
	Generic Notification	
	Other party added	
	Cutof party added	
	INFO3 <b2></b2>	
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required	
	CPG	
	Generic Notification	
	conference established	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)	
Establish a confirmed session from User A in Network A to user B1 in Network B User A establishes a CONF conversation		
	INFO1(Call-ID A-B1, CPG)	
	← 200 INFO	
Establish a confirmed s	ession from User A in Network A to user B2 in Network B and add to the	
	INFO2(Call-ID A-B2, CPG)	
	INFO3(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>
	INFO3 <b2> CPG Generic Notification= reattached</b2>
	INFO4 <b2></b2>

Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
` `	a CONF com	munication with User B1 and Us	` ,
	User A isola	tes User B1 from the CONF conv	versation
		INFO1(Call-ID A-B1, CPG)	→
	←	200 INFO	
		INFO3(Call-ID A-B2, CPG)	→
	←	200 INFO	
	User A reatta	aches User B1 to the CONF conv	versation
		INFO2(Call-ID A-B2, CPG)	→
	←	200 INFO	
		INFO4(Call-ID A-B2, CPG)	→
	←	200 INFO	
		Apply post test routine	
Comments	User A	nvokes a CONF conversation with	User B1 and User b2 in Network B
	User A s	plits user B1 in Network B from the	e CONF conversation
	Check:	Is an INFO request sent to user E	31 and the Generic notification is set
		to 'isolated' in the encapsulated (CPG?
	Check:	Is an INFO request sent to user E	32 and the Generic notification is set
		to 'Other party isolated' in the end	capsulated CPG?
	User A r	eattaches user B1 in Network B to	the CONF conversation
	Check:	Is an INFO request sent to user E	31 and the Generic notification is set
		to 'reattached' in the encapsulate	ed CPG?
	Check:	Is an INFO request sent to user E	32 and the Generic notification is set
		to 'Other party reattached' in the	encapsulated CPG?
	Repeat t	his test in reverse direction	

Test case number	SS conf 009
	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>
	INFO2 <b1> CPG Generic Notification=Other party split</b1>
	INFO3 <b2> CPG Generic Notification=Conference established</b2>
	INFO4 <b2> CPG Generic Notification= Other party added</b2>
Message flow	Control Hounteducti — Outer party added
SIP (Network A) Establish a	Interconnection Interface SIP (Network B) CONF communication with User B1 and User B2 in Network B ser A isolates User B1 from the CONF conversation
	INFO1(Call-ID A-B1, CPG) → 200 INFO INFO3(Call-ID A-B2, CPG) →
112	INFO3(Call-ID A-B2, CPG) → € 200 INFO ser A reattaches User B1 to the CONF conversation
Us	INFO2(Call-ID A-B2, CPG) 200 INFO
	INFO4(Call-ID A-B2, CPG) → 200 INFO CPG
	Apply post test routine
Comments	User A Invokes a CONF conversation with User B1 and User b2 in Network B User A splits user B1 in Network B from the CONF conversation. Check: Is an INFO request sent to user B1 and the Generic notification is set
	to 'conference disconnected' in the encapsulated CPG? Check: Is an INFO request sent to user B2 and the Generic notification is set to 'Other party split' in the encapsulated CPG? User A adds user B1 in Network B to the CONF conversation.
	Check: Is an INFO request sent to user B1 and the Generic notification is set to 'Conference established' in the encapsulated CPG? Check: Is an INFO request sent to user B2 and the Generic notification is set
	to 'Other party added' in the encapsulated CPG? Repeat this test in reverse direction

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

Test case number	SS_acr-	cb 001			
Test case group		SIP-SIP/Service/ACR-CB			
Reference		4.5.2.6/ [12]			
SELECTION EXPRESSION	SE 32	[12]			
Test purpose		Call Barring performed in network B for user B.			
rest purpose	Can Bar	ring periorined in network B it	or user b.		
	User A is	User A is located in network A and user B is located in network B and is			
		subscribed to the Incoming Call Barring service.			
		Ensure that a communication from user A is rejected in network B by sending a			
		603 Decline due to the Call Barring service of user B.			
Configuration		User B is subscribed to the incoming Call Barring service (e.g. user A in a black			
3	list)	3	3	. 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:	Check: Is the communication rejected by sending a 603 (Decline) final			
	response	e 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</uri>	
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause = 21	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 603 Decline (REL)	
	ACK →	
Comments	Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the communication rejected by sending a 603 Decline final response sent to user A? Check: Is an ISUP/BICC REL is present in the 603 and the cause value is set to '21'?	
	Repeat this test in reverse direction.	

7.1.5.9 Closed User Group (CUG)

Test case number	SS_cug_	001	
Test case group	SIP-SIP/	Service/CUG	
Reference	4.5.2.4/ [13]		
SELECTION EXPRESSION	SE 34		
Test purpose	Originating user +OA to terminating user no CUG.		
	An originating user in a CUG Outgoing Access allowed calls to a user not in a		
	CUG. The session establishment is successful.		
Configuration	Originat	ing user: CUG, outgoing access allowed	
SIP Parameter	INVITE:		
	Content-Type: application/vnd.etsi.cug+xml		
	Conte	ent-Disposition: signal;handling=	
	<cug> <networkindicator>01 <networkindicator>23 <cuginterlockbinarycode>0F03 <cugcommunicationindicator>10 <cugcommunicationindicator>10</cugcommunicationindicator></cugcommunicationindicator></cuginterlockbinarycode></networkindicator></networkindicator></cug>		
Managara flavo	<:C	ug>	
Message flow SIP (Network A)		Interconnection Interface SIP (Network B)	
SIP (Network A)		INVITE -	
	←	180 Ringing	
	•	Apply post test routine	
Comments	Check:	Is the Content-Type in The INVITE set to	
Comments	Oncok.	application/vnd.etsi.cug+xml?	
	Check:	Contains the XML body in the INVITE a 'cug' element?	
	Check:	Contains the XML body in the INVITE a 'networkIndicator' element as	
		a 'cug' child element?	
	Check:	Contains the XML body in the INVITE a 'cugInterlockBinaryCode'	
		element as a 'cug' child element?	
	Check:	Contains the XML body in the INVITE a 'cugCommunicationIndicator'	
	element set to '10' as a 'cug' child element?		
	Check: Is the session setup not rejected?		
	Repeat tl	nis test in reverse direction.	

Test case number	SS_cug_	002
Test case group		Service/CUG
Reference	1	1.5.2.10/ [13]
SELECTION EXPRESSION	SE 34	
Test purpose		ing user -OA to terminating user no CUG.
rest purpose	Originati	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
		The session establishment is not successful, a 403 (Forbidden)
	response	,
Configuration		ing user: CUG, outgoing access not allowed
SIP Parameter	INVITE:	3
		ent-Type: application/vnd.etsi.cug+xml
		ent-Disposition: signal;handling= required
	<:C	ug>
	<networkindicator>01</networkindicator>	
	<.	networkIndicator>23
		ugInterlockBinaryCode>0F03
	<.	cugCommunicationIndicator>11
	<cug></cug>	
Message flow		
SIP (Network A)		Interconnection Interface SIP (Network B)
	_	INVITE →
	←	403 (Forbidden)
		ACK →
Comments	Check:	Is the Content-Type in The INVITE set to
		application/vnd.etsi.cug+xml?
	Check:	Is the handling parameter in the Content-Disposition header set to
	01	required?
	Check:	Contains the XML body in the INVITE a 'cug' element?
	Check:	Contains the XML body in the INVITE a 'networkIndicator' element as
	01	a 'cug' child element?
	Check:	Contains the XML body in the INVITE a 'cugInterlockBinaryCode'
	Chaak	element as a 'cug' child element?
	Check:	Contains the XML body in the INVITE a 'cugCommunicationIndicator'
	Check:	element set to '11' as a 'cug' child element? Is the session setup rejected? A 403 (Forbidden) final response is sent
	Check.	
	Donoct ti	by the terminating network?
	NOTE:	nis test in reverse direction. The networkIndicator element value and the cugInterlockBinaryCode
	NOTE.	element value are examples.

Test case number	SS_cug_003
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.4, 4.5.2.10/ [13]
SELECTION EXPRESSION	SE 34
Test purpose	Originating user -OA to terminating user -IA.
	gramming account to the control of t
	An originating user in a CUG Outgoing Access not allowed calls to a user in the
	same CUG Incoming Access not allowed. The session establishment is
	successful.
Configuration	Originating user: CUG, outgoing access not allowed
	Terminating user: CUG incoming access not allowed
	User in network A and user in network B are in the same CUG
SIP Parameter	INVITE:
	Content-Type: application/vnd.etsi.cug+xml
	Content-Disposition: signal;handling= required
	<cug></cug>
	<networkindicator>01</networkindicator> 23
	<retworkindicator>23<cuginterlockbinarycode>0F03</cuginterlockbinarycode></retworkindicator>
	<cug interlockbinarycode="">or-os</cug>
	<cug></cug>
Message flow	- Children
SIP (Network A)	Interconnection Interface SIP (Network B)
, ,	INVITE →
	← 180 Ringing
	Apply post test routine
Comments	Check: Is the Content-Type in The INVITE set to
	application/vnd.etsi.cug+xml?
	Check: Is the handling parameter in the Content-Disposition header set to
	required?
	Check: Contains the XML body in the INVITE a 'cug' element?
	Check: Contains the XML body in the INVITE a 'networkIndicator' element as
	a 'cug' child element?
	Check: Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element?
	Check: Contains the XML body in the INVITE a 'cugCommunicationIndicator'
	element set to '11' as a 'cug' child element?
	Check: Is the session setup not rejected?
	Repeat this test in reverse direction.
	NOTE: The networkIndicator element value and the cugInterlockBinaryCode
	1.12 1 1.10 Hotel of the land of the distriction of the land of the degrit of the land
	element value are examples.

Test case number	SS_cug_	004
Test case group		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.
		nating user in a CUG calls to a user in a different CUG Incoming Access wed. The session establishment is not successful, a 403 (Forbidden) e is sent.
Configuration		network A and user in network B are not in the same CUG
	Termina	ting user: CUG incoming access not allowed
SIP Parameter		ent-Type: application/vnd.etsi.cug+xml ent-Disposition: signal;handling=
	< <	networkIndicator>01230F03cugCommunicationIndicator>
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? 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'
	Check:	element as a 'cug' child element? Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '10' or '11'as a 'cug' child element?
	Check:	Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network?
	Repeat t	his test in reverse direction.
	NOTE:	The networkIndicator element value and the cugInterlockBinaryCode
		element value are examples.

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

Test case number	SS_cug_	007	
Test case group		Service/CUG	
Reference	4.5.2.4/		
SELECTION EXPRESSION	SE 34	10]	
Test purpose		ing user -OA, network B does not support CUG.	
l'est pui pose	Originati	ing user -OA, network B does not support CoG.	
	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:		
	Conte	ent-Type: application/vnd.etsi.cug+xml	
	Conte	ent-Disposition: signal;handling= required	
	<cug></cug>		
	1	networkIndicator>01	
	<networkindicator>23</networkindicator>		
		cugInterlockBinaryCode>0F03	
	<cugcommunicationindicator>11</cugcommunicationindicator>		
Message flow	\c		
SIP (Network A)		Interconnection Interface SIP (Network B) INVITE →	
	←	4xx/501 Not Implemented	
	•	ACK →	
Comments	Check:	Is the Content-Type in The INVITE set to	
		application/vnd.etsi.cug+xml?	
	Check:	Is the handling parameter in the Content-Disposition header set to	
		required?	
	Check:	Contains the XML body in the INVITE a 'cug' element?	
	Check:	Contains the XML body in the INVITE a 'networkIndicator' element as	
		a 'cug' child element?	
	Check:	Contains the XML body in the INVITE a 'cugInterlockBinaryCode'	
		element as a 'cug' child element?	
	Check:	Contains the XML body in the INVITE a 'cugCommunicationIndicator'	
		element set to '11' as a 'cug' child element?	
	Check:	Is the session setup rejected by sending an unsuccessful final	
	Donoot #	response?	
	NOTE:	nis test in reverse direction. The networkIndicator element value and the cugInterlockBinaryCode	
	INUIE.	The networkingicator element value and the cugintenockbinarycode	
		element value are examples.	

Test case number	SS_cug_	007A
Test case group		Service/CUG
Reference	4.5.2.4/	
SELECTION EXPRESSION	SE 34	10]
Test purpose		ing user CUG-OA to terminating CUG user +ICB
rest purpose	Originat	ing user 000-0A to terminating 000 user FIOD
	An origin	nating user in a CUG outgoing access not allowed calls to a user in the
		JG Incoming communication barred. The session establishment is not
		rul, a 603 (Decline) response is sent.
Configuration		Network B in a CUG incoming Communication Barring
SIP Parameter	INVITE:	tetwork B in a coc incoming communication Barring
On Tarameter		ent-Type: application/vnd.etsi.cug+xml
		ent-Type: application/vird.etsl.oug+xilli ent-Disposition: signal;handling= required
	Cont	ent-Disposition. signal, nationing—required
	<ci< th=""><th></th></ci<>	
		networkIndicator>01
		networkIndicator>23
		cugInterlockBinaryCode>0F03
		cugCommunicationIndicator>11
	<cug></cug>	
Message flow	· L	
SIP (Network A)		Interconnection Interface SIP (Network B)
		INVITE →
	←	603 Decline
		ACK →
Comments	Check:	Is the Content-Type in The INVITE set to
		application/vnd.etsi.cug+xml?
	Check:	Is the handling parameter in the Content-Disposition header set to
		required?
	Check:	Contains the XML body in the INVITE a 'cug' element?
	Check:	Contains the XML body in the INVITE a 'networkIndicator' element as
		a 'cug' child element?
	Check:	Contains the XML body in the INVITE a 'cugInterlockBinaryCode'
		element as a 'cug' child element?
	Check:	Contains the XML body in the INVITE a 'cugCommunicationIndicator'
		element set to '11' as a 'cug' child element?
	Check:	element set to '11' as a 'cug' child element? Is the session setup rejected by sending a 603 Decline unsuccessful
	Check:	element set to '11' as a 'cug' child element? Is the session setup rejected by sending a 603 Decline unsuccessful final response?
	Check:	element set to '11' as a 'cug' child element? Is the session setup rejected by sending a 603 Decline unsuccessful final response? his test in reverse direction.
	Check:	element set to '11' as a 'cug' child element? Is the session setup rejected by sending a 603 Decline unsuccessful final response?

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

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

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 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 User A and User B are in the same CUG
SIP Parameter	INVITE
	Content-Type: multipart/mixed;boundary=[any boundary name][any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required
	IAM
	Optional Forward call indicator CUG Call Indicator Outgoing access not allowed CUG interlock code [any boundary name]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE →
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.
	Interest this test in reverse uncontrol.

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

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 allowed.
	User A is located in Network A. User B is located in the F SIP-I - ISUP/BICC interworking applied. Ensure that whe B Incoming access allowed in Network B. The call is succ	n user A calls CUG user
Configuration	 User in PSTN/PLMN part of Network B in a CUG inc 	coming access allowed
SIP Parameter		,
Message flow SIP (Network A)	Interconnection Interface INVITE →	SIP (Network 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/[15]		
SELECTION EXPRESSION	SE 35 AN	ND SE 36		
Test purpose	Call reje	cted after timeout TAS-CW.		
	the comm Ensure the Waiting in or UDUB	s located in network A, user B is lo nunication Waiting service. nat when user A calls user B, user ndication' in the 180 Ringing providual. After timeout TAS-CW network Epole) response toward user A and the	· A receives sional resp 3 sends a 4	s the 'communication onse if the user B is NDUB 180 (Temporarily
Configuration				
SIP Parameter	180: Alert-Info: <urn:alert:service:call-waiting> 480:</urn:alert:service:call-waiting>			
	Reas	on: 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: 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 th	nis test in reverse direction.		

Test case number	SS cw 003		
Test case group	SIP-SIP/Service/CW		
Reference	6.5/ [24]		
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 59		
Test purpose	SIP-I support. Call Waiting indication in 180 with encapsulated ACM.		
	User A is located in network A, user B is located in the PSTN/PLMN part of		
	network B and subscribed to the Call Waiting service.		
	Ensure that when user A calls user B, an encapsulated ISUP/BICC ACM		
	Generic notification 'call is a waiting call' is present in the 180 Ringing provisional		
	response if the user B is NDUB.		
Configuration	User B subscribed to the CW service		
SIP Parameter	180		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	ACM		
	Backward call indicator		
	Called party's status indicator		
	subscriber free		
	Generic notification		
	Notification indicator		
	call is a waiting call		
Message flow	•		
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE →		
	← 180 Ringing		
	Apply post test routine		
Comments	Check: Is an ISUP/BICC ACM present in the 180 provisional response and		
	the Generic notification is set to 'call is a waiting cal'?		
	Repeat this test in reverse direction.		

Test case number	SS cw 004		
Test case group	SIP-SIP/Service/CW		
Reference	6.5/ [24]		
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 59		
Test purpose	SIP-I support. Call Waiting indication in 180 with encapsulated CPG.		
	3		
	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 information Event indicator		
	ALERTING		
	Generic notification		
	Notification indicator		
	call is a waiting call		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
,	INVITE →		
	← 183 Session Progress (ACM)		
	← 180 Ringing (CPG)		
	Apply post test routine		
Comments	Check: Is an ISUP/BICC CPG present in the 180 provisional response and the		
	Generic notification is set to 'call is a waiting cal'?		
	Repeat this test in reverse direction.		

7.1.5.11 Explicit Communication Transfer (ECT)

Test case number	SS_ect_001		
Test case group	SIP-SIP/Service/ECT		
Reference	4.5.2/ [11]		
SELECTION EXPRESSION	[Network A] SE 37 AND [Network A] SE 11 AND [Network A] SE 49		
Test purpose	Blind/assured transfer using the REFER method.		
	-		
	User A is located in network A, user B and user C are located in network B. User		
	A invokes ECT to transfer a session with user B to user C.		
	 Ensure that a REFER request is sent from network A to network B in the 		
	dialogue with user B. The URI in the Refer-To header is set to the		
	address of the ECT AS in network A and the method parameter is set to		
	'INVITE'.		
	 Ensure that an INVITE request is sent from network B to network A and 		
	the Request URI is set to the address of the ECT AS in network A.		
	Ensure that an INVITE request is sent from network A to network B and		
	the Request URI is set to the address of user C.		
Configuration	DEFEN D. CHIDLE I. C. D.		
SIP Parameter	REFER: Request URI address of user B Refer-To: <uri ect-as="" of="">; method=invite</uri>		
	Refer-10. <uri ec1-a5="" of="">, Method=Invite</uri>		
	INVITE1 Request URI address of ECT-AS		
	TVITE I Request our address of EoT Ao		
	INVITE2: Request URI address of user C		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
A confirm	ned session is established between user A and user B		
	ned session is established between user A and user C		
Use	User A invokes ECT to transfer the session to user C		
	REFER →		
	202 Accepted		
	NOTIFY (100)		
	200 OK NOTIFY →		
CAS	E Blind transfer		
CAC	BYE (A-B)		
	€ 200 OK BYE		
	200 SINDIE		
	← INVITE 1 (ECT-AS)		
	INVITE2 (user C) →		
	€ 200 OK ÎNVITE		
	ACK →		
	200 OK INVITE →		
	← ACK		
	NOTIFY (200)		
	200 OK NOTIFY →		
0455	Accuract transfer		
CASE	Assured transfer		
	BYE (A-B) → 200 OK BYE		
	Apply post test routine		
	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:	
	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_ect_002		
Test case group	SIP-SIP/Service/ECT		
Reference	4.5.2/ [11]		
SELECTION EXPRESSION	[Network A] SE37 AND [Network A] SE 11 AND [Network A] SE 50		
Test purpose	Consultative transfer using the REFER method.		
Configuration SIP Parameter	User A is located in network A, user B and user C are located in network B. User A invokes ECT to transfer a session with user B to user C. • Ensure that a REFER request is sent from network A to network B in the dialogue with user B. The URI in the Refer-To header is set to the address of the ECT AS in network A and the method parameter is set to 'INVITE'. • Ensure that an INVITE request is sent from network B to network A and the Request URI is set to the address of the ECT AS in network A. • Ensure that an INVITE request is sent from network A to network B and the Request URI is set to the address of user C and a Replaces header is present containing the session identifiers of the session A - C. REFER:Request URI address of user B Refer-To: <uri ect-as="" of="">; method=invite</uri>		
	INVITE2: Request URI address of user C		
	Require: replaces		
Message flow	Replaces: <session a-c=""></session>		
	rmed session is established between user A and user C er A invokes ECT to transfer the session to user C REFER 202 Accepted		
	← NOTIFY (100) 200 OK NOTIFY →		
	 INVITE 1 (ECT-AS) INVITE 2 (user C) 200 OK INVITE ACK 		
	200 OK INVITE → ← ACK ← NOTIFY (200) 200 OK NOTIFY →		
	BYE (A-B) → 200 OK BYE		
	← BYE (A-C) 200 OK BYE → Apply post test routine		
Comments	Check: Is a REFER request is sent network B, the Refer-To header is set to the URI of the ECT-AS in network A and a method parameter is present set to 'INVITE'?		
	Check: Is an INVITE request sent to network A the Request line is set to the		
	address of the ECT-AS in network A? Check: Is an INVITE request is sent to network B the Request is set to the address of user C and a Replaces header is present contains the session identifiers of the session A-C?		
	Check: Is the session A - B and the session A - C terminated? Check: Ensure the property of speech between user B and user C. Repeat this test in reverse direction.		

Test case number	SS_ect (003		
Test case group		SIP-SIP/Service/ECT		
Reference		4.5.2/ [11], 4.7.2.9.7/ [20]		
SELECTION EXPRESSION		A] SE37 AND NOT [Network A]	SE 12 AND [Network A] SE 49	
Test purpose		sured transfer using the 3pcc r		
		.		
	User A is	located in network A, user B an	user C are located in network B User A	
	invokes E	ECT to transfer a session with use	er B to user C.	
	_	Ensure that the network A establi		
	• E	Ensure that the network A sends	a reINVITE to update the session	
	t	petween user A and user B (SDP	: IP address, port and codec).	
Configuration				
SIP Parameter	INVITE1	Request URI address of user C		
	INIV/ITEO	Danisat IIDI addasa at was D		
	SDP	Request URI address of user B		
	_	IN IP4/6 [new IP address]		
			codec list1	
		m=audio [new port] RTP/AVP [new codec list] a=[new attributes]		
Message flow	<u> </u>	-[now attributed]		
SIP (Network A)		Interconnection Interface	SIP (Network B)	
	firmed sess	ion is established between use		
U	lser A invok	es ECT to transfer the session	to user C	
		INVITE1 (user C)	→	
	←	180 Ringing		
	←	200 OK INVITE		
		ACK	→	
		INIVITED (WAST D)	→	
	←	INVITE2 (user B) 200 OK INVITE	7	
	•	ACK	→	
		Apply post test routine	•	
Comments	Check:		network A to user C to establish a	
		dialogue between network A an		
	Check:			
		parameter in the SDP?		
	Repeat th	nis test in reverse direction.		
	•			

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 is located in hetwork A, user B and user C are located in hetwork B		
	Ensure that the network A sends a reINVITE to update the session hot was a vess A and vess B (CDB III) address part and eache		
	between user A and user B (SDP: IP address, port and codec).		
	Ensure that the network A sends a reINVITE to update the session between year A and year C (SDR II) address, part and cades)		
Configuration	between user A and user C (SDP: IP address, port and codec).		
Configuration	INVITEA Degreet 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 attributes]		
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	205		
Test case group	SIP-SIP/Service/ECT			
Reference				
SELECTION EXPRESSION	5.4.3.2/ [24] [Network A] SE 17 AND SE 47 AND SE 60			
	SIP-I support. Call Transfer invoked in active state, call was previous on			
Test purpose	HOLD. BICC/ISUP - SIP-I interworking applies in the originating network User A and C are located in network A and user B is located in network B.			
		Ensure that an User A can successfully invoke the ECT supplementary service and transfer the call with User B to User C in active state.		
Configuration			Transfer supplementary service	
SIP Parameter	INVITE	Subscribed to the Explicit earl	Transier supplementary service	
on ruramotor	Co	ontent-Type: multipart/mixed;bo	undary=[any boundary name]	
	Co a=	ontent-Type: application/sdp =sendrecv		
	Co Co	any boundary name] ontent-Type: application/isup;ve ontent-Disposition: signal;handli AC		
		Generic Notification Call transfer active		
		Call transfer number		
	[any boundary name]		
		Interconnection Interface stablished between user A and tes ECT to transfer the session INFO (LOP request)		
	←	200 OK INFO	7	
	~	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	→	
	1.	Apply post test routine		
Comments	User A se	n from User A to User B is alrea ets the User B on hold wokes the ECT service		
	Check:	and an ISUP LOP message is set to 'request'?	is sent from Network A to Network B present the Loop prevention indicator	
	Check:		is sent from Network A to Network B present the Loop prevention indicator	
	Check:		st sent and an ISUP FAC message is notification indicator is set to 'Call the media stream is set to	
	Check:			
	NOTE:	The content of the FAC in the of the FAC in the INFO reques	INVITE request is Equal to the content	
	Repeat th	nis test in reverse direction.		

Test case number	SS_ect_0	006	
Test case group	SIP-SIP/Service/ECT		
Reference	5.4.3.2/ [24]		
SELECTION EXPRESSION		A] SE 17 AND SE 47 AND SE	60
Test purpose			alerting state, call was previous on
	HOLD.		
			n the originating network User A and C
		ed in network A and user B is lo	
			nvoke the ECT supplementary service
0		sfer the call with User B to User	
Configuration		s subscribed to the Explicit Call	ransfer supplementary service
SIP Parameter	INVITE C	ontent-Type: multipart/mixed;bo	undary=[any boundary name]
		[any boundary name]	
	C	ontent-Type: application/sdp	
	a=	=sendrecv	
	[[any boundary name]	
		ontent-Type: application/isup;ve	rsion-itu-t92
		ontent-Type: application/isup,ve ontent-Disposition: signal;handli	
		AC	ng-roquirou
		Generic Notification	
		Call transfer alerting	
		Call transfer number	
		[any boundary name]	
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
		stablished between user A and	
Us	ser A invok	ces ECT to transfer the sessio	
	_	INFO (LOP request)	→
	(200 OK INFO	
	←	INFO (LOP response)	•
CASE A		200 OK INFO	→
CASE A		INVITE (sendrecv; FAC)	→
	←	200 OK INVITE	7
	•	ACK	→
CASE B		non	•
0.1022		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
		nvokes the ECT service	to constitue as New York No. 1 P.
	Check:		is sent from Network A to Network B
			present the Loop prevention indicator
	Charles	set to 'request'?	is contifrom Notwork A to National D
	Check:		is sent from Network A to Network B
		set to 'response'?	present the Loop prevention indicator
	Check:		st sent and an ISUP FAC message is
	J.IOOK.		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 t	his test in reverse direction.	
μ			

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 purpose	on Tsupport. Oan Transier invoked in delive 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	
on randicier	Content-Type: application/isup;version=itu-t92	
	Content-Type: application/isup,version=itu-to2 Content-Disposition: signal;handling=required	
	FAC	
	Generic Notification	
	Call transfer active	
	Call transfer number	
Message flow	Can danier Harrison	
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.	
	DIOC/IOUD CID Linterwooding and in the administration material Linear A and D	
	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	
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 (CPG) →	
	← 200 OK INFO	
	Apply post test routine	
Comments	A session from User A to User B is already established	
	User A sets the User B on hold	
	A session from User A to User C is already established	
	User A invokes the ECT service	
	Check: Is (optional) an INFO request is sent from Network A to Network B	
	and an ISUP LOP message is present the Loop prevention indicator	
	set to 'request'?	
	Check: Is (optional) an INFO request is sent from Network A to Network B	
	and an ISUP LOP message is present the Loop prevention indicator	
	set to 'response'?	
	Check: Is (CASE B) an INFO request sent and an ISUP CPG message is	
	present containing a Generic notification indicator is set to 'Call	
	transfer alerting'?	
	NOTE: The content of the FAC in the INVITE request is Equal to the content	
	of the FAC in the INFO request.	
	Repeat this test in reverse direction.	

7.1.5.12 Malicious Communication Identification (MCID)

Test case number	SS_mcid	SS_mcid_001		
Test case group	SIP-SIP/	SIP-SIP/Service/MCID		
Reference	4.5.2.5/[18]		
SELECTION EXPRESSION	SE 38			
Test purpose	Network	B sends a MCID request, no re	esponse.	
	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			
		esponse.	TO ID the net	WORK D SCHOOL THE 100
Configuration		s subscribed to the MCID service		
SIP Parameter	INFO:	o cascerised to the Mers corvies		
	<:mcid> <:request> <:McidRequestIndicator>01 :McidRequestIndicator <:HoldingIndicator > :HoldingIndicator			
		/:request>		
Message flow	:mcid			
SIP (Network A)	←	Interconnection Interface INVITE INFO	→	SIP (Network B)
		200 OK INFO	→	
	_	Timeout T _{O-ID}		
	←	180 Ringing		
		Apply post test routine	1.40	
Comments	Check:	Is an INFO request sent to net		2412
	Check:	Is the McidRequestIndicator ele		
	Check:	is a 200 OK INFO response se	ni io network	D!
	Repeat	his test in reverse direction.		

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

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 a 183 Session Progress to network A and an ISUP/BICC IDR message is present the MCID request indicator is set to 'MCID requested' requesting the originating identity. After timeout of timer (ISUP) T39 the network B sends the 180 Ringing response.		
Configuration	User B is subscribed to the MCID service		
SIP Parameter	INFO:		
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IDR MCID request indicators MCID request indicator MCID requested		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 183 Session Progress(IDR) Timeout T _{O-ID}		
	← 180 Ringing		
	Apply post test routine		
Comments	Check: Is an INFO request sent to network A?		
	Check: Is a ISUP/BICC IDR message is present and the MCID request indicator is set to 'MCID requested'?		
	Check: Is a 200 OK INFO response sent to network B?		
	NOTE: Based on network policies the MCID request indicator can be set		
	to'MCID not requested'.		
	Repeat this test in reverse direction.		
	Tropodi uno toti in fovoro 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 is sented a more request, more 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 a INFO request to network A 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		
Comigaration	User A is a ISDN or POTS user in the PSTN of network A		
SIP Parameter	INFO:		
on rarameter	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	Content-Disposition. Signal, nariding-required		
	IDR		
	MCID request indicators		
	MCID request indicator		
	MCID requested		
	Molb requested		
	INFO:		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	Softon Bioposition: signati, narialing roquired		
	IRS		
	MCID response indicators		
	MCID response indicator		
	MCID included		
	Calling party number		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
,	INVITE -		
	← INFO(IDR)		
	200 OK INFO →		
	INFO(IRS) →		
	€ 200 OK INFO		
	← 180 Ringing		
	Apply post test routine		
Comments	Check: Is an INFO request sent to network A and a ISUP/BICC IDR is present		
	and the MCID request indicator is set to 'MCID requested'?		
	Check: Is a 200 OK INFO response sent to network B?		
	Check: Is an INFO request sent to network B and a ISUP/BICC IRS is present		
	and the MCID response indicator is set to 'MCID included'?		
	Check: Is the Calling party number present in the attached ISUP/BICC IRS?		
	Check: Is a 200 OK INFO response sent to network A?		
	Repeat this test in reverse direction.		
	proposition to the control of the co		

7.1.5.13 Message Waiting Indication (MWI)

Test case number	SS_r	SS_mwi_001		
Test case group	SIP-S	SIP-SIP/Service/MWI		
Reference	4.7.2	4.7.2/ [16]		
SELECTION EXPRESSION	Netv	[Network A] SE 39 AND [Network B] SE 39		
Test purpose		al subscription of a Voicemail box.		
		•		
			Voicemail box is located in network B.	
		re that a Voicemail owner is able to a	activate his Voicemail box.	
Configuration	Voice	email in network B		
	Voice	email owner in network A		
SIP Parameter	SUB	CRIBE		
		Event: message-summary		
		Expires: [any value]		
		Accept: application/simple-mes	sage-summary	
	NOT			
		Subscription-State: active;expir	es=[any value]	
Managara di sur		Event: message-summary		
Message flow SIP (Network A)		Interconnection Interface	SIP (Network B)	
SIF (Network A)		SUBCRIBE	⇒	
	←	200 OK SUBSCRIBE	,	
	•	200 OK GODGONIBE		
	←	NOTIFY		
		200 OK NOTIFY	→	
	←	200 OK BYE		
	←	NOTIFY		
		200 OK NOTIFY	→	
	•	Apply post test routine		
Comments	Check:		rk A to subscribe to a Voicemail box in	
		network B?		
	Check:		CRIBE set to 'message-summary'?	
	Check:		BCRIBE set to 'application/simple-	
	Chaola	message-summary'?	EV is not to Impagage aummoral?	
	Check:		FY is set to 'message-summary'?	
	Repeat	this test in reverse direction.		

Test case number	SS_mwi_002		
Test case group	SIP-SIP/Service/MWI		
Reference	4.7.2/ [16]		
SELECTION EXPRESSION	[Network A] SE 39 AND [Network B] SE 39		
Test purpose	A new entry in the Voicemail box is indicated to the owner.		
	The Voicemail owner is in network A, his Voicemail box is located in network B. Ensure when a user calls user A and the call is not answered, the call is forwarded to the Voicemail box of user A in network B. Ensure that the user A is notified by message waiting indication that there is a new message present in his voicemail account.		
Configuration	Voicemail in network B		
SIP Parameter	Voicemail owner in network A 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/simplemessage-summary'?		
	Check: Contains the MIME body the header 'Messages-Waiting' set to 'yes'? Check: Contains the MIME body the optional header 'Message-Account'? Check: Contains the MIME body the optional header 'Voice-Message'? Repeat this test in reverse direction.		

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

Test case number	SS_cc_0	01	
Test case group	SIP-SIP/	Service/CC	
Reference	4.5.4.3/		
SELECTION EXPRESSION		A] SE 40 AND [Network B] SE 40	
Test purpose		ng of CCBS possible.	
root parpood	a.oa	.g c. cc2c peccisio.	
	User A is	s located in network A and user B is	s located in network B
		when user A calls user B and user E	
		that CCBS is possible in the 486	•
Configuration		a. 3323 to possible	240) 11010 111141 1000011001
SIP Parameter	486:		
		nfo: <sip:ue-b>:purpose=call-com</sip:ue-b>	pletion:m=BS
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
Cii (iiciiiciii)		INVITE	→
	←	486 Busy Here	
		ACK	→
Comments	Check:	The 486 final response contains t	the Call-Info header.
	Check:		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 param	neter set to 'BS'.
	Repeat t	his 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	
SELECTION EXPRESSION	4.5.4.2/ [14] ([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR
SELECTION EXPRESSION	
Toot nurnoon	[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	and request is in the Gail completion quode at the terminating rippheation cerver.
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.

Test case number	SS cc 004	
Test case group	SIP-SIP/Service	ce/CC
Reference	4.5.4.3/ [14]	
SELECTION EXPRESSION	([Network A] S	E 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR
	[Network B] SE	E 41)
Test purpose	Invocation of	CCBS or CCNR unsuccessful; short term denial
	User A is locat	ed in network A and user B is located in network B.
	network B is cu	er A invokes a CCBS or CCNR request to network B and the urrently unable to process the request (e.g. the B-queue is full), a ly Unavailable final response is sent.
Configuration		
SIP Parameter		o:B-AS;m=BS or m=NR
	From: <ue-< th=""><th>• •</th></ue-<>	• •
	To: <ue-b></ue-b>	
	Contact: <a< th=""><th></th></a<>	
	Event:call-o	completion
Message flow		
SIP (Network A)		erconnection Interface SIP (Network B)
An indicatio	whether CCE	3S or CCNR is possible is sent by network B SUBSCRIBE →
	← 480	-
Comments		(Temporarily Unavailable)
Comments		SUBCRIBE request is sent to network B? e m parameter in the Request URI is set to 'BS' in case of CCBS
		est or set to 'NR' in case of CCNR?
		480 Temporarily Unavailable sent from network B indicates the
		BS or CCNR request is unsuccessful e.g. CC queue is full?
		st in reverse direction.
	topout tino tec	ot iii 1070100 diilottolli.

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	WOTEN : 0.40
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	Oubscription otate. terminated, reason=noresource
SIP (Network A)	Interconnection Interface SIP (Network B)
	CCBS or CCNR request was already successful
	← NOTIFY
	200 OK NOTIFY →
	INVITE ->
	← 180 Ringing
	← NOTIFY
	200 OK NOTIFY →
	200 OK NOTIFT - 7
	€ 200 OK INVITE
	ACK →
	Apply post test routine
Comments	Check: Is a NOTIFY request is sent to network A and the Event header is set
	to 'call-completion' and the state header in the message body is set to
	'ready'?
	Check: Is the recall from user A to user B is successful?
	Check: Is the CC revocation is performed after the 180 Ringing or the 200 OK
	INVITE was sent to user A?
	In the contract of the contrac
	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 A CCBS or CCNR request was already successful User B is available for recall		
	← NOTIFY	
	200 OK NOTIFY →	
	CC-T9 expires	
	← NOTIFY	
Comments	200 OK NOTIFY → 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'?	
	User A does not perform the recall	
	Check: Is the CC revocation is performed after timer CC-T9 expires?	
	Repeat this test in reverse direction.	

Test case number	SS_cc_007		
Test case group	SIP-SIP/Service/CC		
Reference	4.5.4.2/ [14]		
SELECTION EXPRESSION	([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR		
	[Network B] SE 41)		
Test purpose	User A is unavailable while CC recall is performed.		
	·		
	User A is located in network A and user B is located in network B. User A has		
	successfully invoked a CCBS or CCNR request. User B is available for CC-recall		
	and network B sends a CC-recall notification to network A.		
	Ensure that network A sends PUBLISH request to suspend the recall		
	procedure.		
	Ensure that network A sends PUBLISH request to resume the recall		
	procedure if user A is available to complete the recall procedure.		
	Ensure the network B sends a NOTIFY request to indicate the CC-recall procedure.		
Configuration	procedure.		
SIP Parameter	NOTIFY sip:O-AS		
Sir raiametei	Event:call-completion		
	Content-Type: application/call-completion		
	state: ready		
	PUBLISH sip B-AS		
	To: SIP 2		
	Event: presence		
	Content-Type: application/pidf+xml		
	xml version="1.0" encoding="UTF-8"?		
	<pre><pre><pre><pre></pre></pre></pre></pre>		
	<status></status>		
	<basic>closed</basic>		
	PUBLISH sip B-AS		
	To: SIP 2		
	Event: presence		
	Content-Type: application/pidf+xml		
	xml version="1.0" encoding="UTF-8"?		
	<pre><pre><pre><pre><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 →		
Comments	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 SIP (Network A)	Interconnection Interface INVITE 486 Busy Here (REL) ACK SIP (Network B) →
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 Signalling (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 and User-to-user Information parameter is present in the
	encapsulated IAM of the initial INVTE request.
Configuration	User A is subscribed to the User-to-User service 1 implicit request
SIP Parameter	INVITE:
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	IAM
	User-to-user Information
	User Information
Message flow	lateres are established as a CID (National D)
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 Information parameter present in the encapsulated ISUP/BICC IAM?
	Repeat this test in reverse direction.

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

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

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		
Message flow	service 1 provided		
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE (IAM) → 180 Ringing (ACM) Apply post test routine		
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request? Check: Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM? Check: Is an ISUP/BICC ACM encapsulated in the 180 response? Check: Is an User-to-user Indicator parameter present set to 'Response', 'service 1 provided' in the encapsulated ISUP/BICC ACM?		
	Repeat this test in reverse direction.		

Test case number	SS_uus_005		
Test case group	SIP-SIP/SIP-I/UUS		
Reference	7.1, 6.5/ [24]		
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63		
Test purpose	SIP-I support: Indicating of User-to-User service 1 not essential explicit rejected in 180.		
	BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 1 explicit request calls user B not subscribed to User-to-User service 1 the call is rejected by the network an User-to-user Indicator parameter is present set to 'Response', 'service 1 not provided' in the encapsulated ACM of the 180 response.		
Configuration	User A is subscribed to the User-to-User service 1 explicit request User B is not subscribed to the User-to-User service 1		
SIP Parameter	User B is not subscribed to the User-to-User service 1 INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM User-to-user Indicator Request service 1 not essential 180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM User-to-user Indicator		
	Response service 1 not provided		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE (IAM) → 180 Ringing (ACM)		
	Apply post test routine		
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request? Check: Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM?		
	Check: Is an ISUP/BICC ACM encapsulated in the 180 response? Check: Is an User-to-user Indicator parameter present set to 'Response', 'service 1 not provided' in the encapsulated ISUP/BICC ACM? Page 24 this test in reverse dispetting.		
	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	#25 01 #05			
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?			

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		
	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		
	containing a User-to-user Indicator parameter, and the indicator		
	Request service 2 is set to the value '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			
	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:			
SIF Farameter	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM			
	User-to-user Indicator Request			
	service 2 not essential or 'essential'			
	180			
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required			
	ACM User-to-user Indicator Response			
	service 2 provided			
	INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required			
	USR User-to-user Information User Information			
	183			
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required USR			
	User-to-user Information User Information			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)			
on (Notificially)	INVITE (IAM) →			
	← 180 Ringing (ACM) INFO (USR) →			
	€ 200 OK INFO			
	← 183 Session Progress (USR)			
	Apply post test routine			
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request containing a User-to-user Indicator parameter, and the indicator			
	Request service 2 is set to the value '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 petwork A to network B containing an User-to-user Information			
	network A to network B containing an User-to-user Information parameter? Check: Is an ISUP/BICC USR encapsulated in the 183 response sent from			
	network B to network A containing an User-to-user Information parameter?			
	Repeat this test in reverse direction.			

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

Test case number	SS_uus_010			
Test case group	SIP-SIP/SIP-I/UUS			
Reference	6.11.2, 7.1/ [24]			
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63			
Test purpose	SIP-I support: Indicating of User-to-User service 2 essential rejection.			
rest purpose	Sir-i support: indicating of User-to-User service 2 essential rejection.			
	BICC/ISUP - SIP-I interworking applies in the originating and terminating network			
	User A is located in network A and user B is located in network B.			
	Ensure when user A subscribed to the User-to-User service 2 essential calls			
	user B not subscribed to User-to-User service 2 the call is rejected by the			
	network. A 500 Server Internal Error is sent and an encapsulated ISUP/BICC			
	REL is present, the Cause value is set to #29 or #69.			
Configuration	User A is subscribed to the User-to-User service 2			
SIP Parameter	INVITE:			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	IAM			
	User-to-user Indicator			
	Request			
	service 2			
	essential			
	500			
	Content-Type: application/isup;version=itu-t92			
	Content-Type: application/isdp,version=itd-ts2			
	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			
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.			
	proposition test in reverse unconstitution			

Test case number	SS_uus_011		
Test case group	SIP-SIP/SIP-I/UUS		
Reference	7.1/[24]		
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 63		
Test purpose	SIP-I support: Indicating of User-to-User service 3 in initial INVITE request.		
	BICC/ISUP - SIP-I interworking applies in the originating network User A is		
	located in network A and user B is located in network B.		
	Ensure when user A subscribed to the User-to-User service 3 calls user B an		
	User-to-user Indicator parameter is present set to 'Request service 3', 'not		
	essential' or 'essential' in the encapsulated IAM of the initial INVTE request.		
Configuration	User A is subscribed to the User-to-User service 3		
SIP Parameter	INVITE:		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	IAM		
	User-to-user Indicator		
	Request		
	service 3		
B	not essential or 'essential'		
Message flow	lutana anno attant lutanfa a		
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE (IAM)		
Comments	Apply post test routine Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request		
Comments			
	containing a User-to-user Indicator parameter, and the indicator Request service 3 is set to the value 'not essential' or 'essential'?		
	Repeat this test in reverse direction.		
	nepeat 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 sup success	oport: Indicating of User-to-User service 3 in initial INVITE request ful.		
		JP - SIP-I interworking applies in the originating network User A is 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', 'n essential' or 'essential' in the encapsulated IAM of the initial INVTE request			
		Jser service is successful.		
Configuration		s subscribed to the User-to-User service 3 s subscribed to the User-to-User service 3		
SIP Parameter	INVITE:	s subscribed to the Oser-to-Oser service 3		
on ranneter	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM			
		User-to-user Indicator		
		Request service 3		
		not essential or 'essential'		
		not ossertial or ossertial		
	200 OK			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	ANM Llear to user Indicator			
	User-to-user Indicator Response			
		service 3 provided		
	INFO			
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required USR			
	User-to-user Information User Information			
Message flow				
SIP (Network A)	£	Interconnection Interface SIP (Network B) INVITE (IAM) →		
	-	180 Ringing (ACM) 200 OK INVITE (ANM) ACK →		
		INFO (USR) →		
	←	200 OK INFO		
	←	INFO (USR)		
		200 OK INFO →		
Comments	Chaoki	Apply post test routine		
Comments	Check:	Is an ISUP/BICC IAM encapsulated in the initial INVITE request containing a User-to-user Indicator parameter, and the indicator		
		Request service 3 is set to the value 'not essential' or 'essential'?		
	Check:	Is an ISUP/BICC ANM encapsulated in the 200 OK INVITE and the		
		User-to-user Indicator parameter is set to 'Response', 'service 3		
		provided'?		
	Check:	Is an ISUP/BICC USR encapsulated in the INFO message sent from		
		network A to network B containing an User-to-user Information parameter?		
	Check:	Is an ISUP/BICC USR encapsulated in the INFO message sent from		
	Oncon.	network B to network A containing an User-to-user Information		
		parameter?		
	Repeat t	his 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		
	63		
Test purpose	SIP-I support: Indicating of User-to-User service 3 not essential rejected in		
	200 OK response.		
	·		
	BICC/ISUP - SIP-I interworking applies in the originating and terminating network		
	User A is located in network A and user B is located in network B.		
	Ensure when user A subscribed to the User-to-User service 3 not essential calls		
	user B not subscribed to User-to-User service 3 the call is rejected by the		
	network an User-to-user Indicator parameter is present set to 'Response',		
	'service 3 not provided' in the encapsulated ANM of the 200 OK final response.		
Configuration	User A is subscribed to the User-to-User service 3		
	User B is not subscribed to the User-to-User service 3		
SIP Parameter	INVITE:		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	IAM		
	User-to-user Indicator		
	Request		
	service 3 not essential		
	not essential		
	200 OK		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	ANM		
	User-to-user Indicator		
	Response		
	service 3 not provided		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE (IAM)		
	← 180 Ringing (ACM) ← 200 OK INVITE (ANM)		
	Apply post test routine		
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request?		
	Check: Is a User-to-user Information parameter present in the encapsulated		
	ISUP/BICC IAM set to 'Request', 'service 3' 'not essential'?		
	Check: Is an ISUP/BICC ANM encapsulated in the 200 OK response?		
	Check: Is an User-to-user Indicator parameter present set to 'Response',		
	'service 3 not provided' in the encapsulated ISUP/BICC ANM?		
	Repeat this test in reverse direction.		

Test case number	SS_uus_014		
Test case group	SIP-SIP/SIP-I/UUS		
Reference	6.11.2, 7.1/ [24]		
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE		
	63		
Test purpose	SIP-I support: Indicating of User-to-User service 3 essential rejection.		
	BICC/ISUP - SIP-I interworking applies in the originating and terminating network		
	User A is located in network A and user B is located in network B.		
	Ensure when user A subscribed to the User-to-User service 3 essential calls		
	user B not subscribed to User-to-User service 3 the call is rejected by the		
	network. A 500 Server Internal Error is sent and an encapsulated ISUP/BICC		
	REL is present, the Cause value is set to #29 or #69.		
Configuration	User A is subscribed to the User-to-User service 3		
SIP Parameter	INVITE:		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	IAM		
	User-to-user Indicator		
	Request		
	service 3		
	essential		
	500		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	REL		
	Cause value		
	#29 or #69		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE (IAM) →		
	← 500 Server Internal Error (REL) ACK		
	Apply post test routine		
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request?		
	Check: Is a User-to-user Indicator parameter present in the encapsulated		
	ISUP/BICC IAM set to 'Request', 'service 1', 'essential'?		
	Check: Is an ISUP/BICC REL encapsulated in the 500 response?		
	Check: Is the Cause value set to #29 or #69 in the encapsulated REL?		
	Repeat this test in reverse direction.		

Test case number	SS_uus_015		
Test case group	SIP-SIP/SIP-I/UUS		
Reference	5.4.3.2, 6.5, 7.1/ [24]		
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE		
	63		
Test purpose	SIP-I support: Indicating of User-to-User service 3 during a session is		
	established successful.		
	BICC/ISUP - SIP-I interworking applies in the originating network User A is		
	located in network A and user B is located in network B.		
	Ensure when user A subscribed to the User-to-User service 3 user A is able to		
	request the User-to-User service 3 while the session is established. The User-to-		
	User service is successful.		
Configuration	User A is subscribed to the User-to-User service 3		
OID D	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		
	not observable		
	INFO:		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	FAA		
	Facility indicator		
	user-to-user service		
	User-to-user Indicator		
	Response		
	service 3 provided		
	INFO		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	USR		
	User-to-user Information		
Massaga flow	User Information		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)		
SIF (Network A)	A session is already established		
	INFO (FAR)		
	← 200 OK INFO		
	← INFO (FAA)		
	200 OK INFO →		
	INFO (USR) →		
	← 200 OK INFO		
	← INFO (USR)		
	200 OK INFO →		
	Apply post test routine		

Comments		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.

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		
Message flow	service 3 not provided		
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 houndary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	IAM		
	Access transport		
	Calling party subaddress		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
-	INVITE(IAM) →		
Comments	Establish a call from User A subscribed to the SUB supplementary service to		
	user B		
	Check: Is an ISUP/BICC IAM present in the initial INVITE request?		
	Check: Is an ISUP/BICC ATP parameter present in the encapsulated IAM		
	containing a Calling party subaddress?		
	Repeat this test in reverse direction.		

Test case number	SS sub 002		
Test case group	SIP-SIP/SIP-I/SUB		
Reference	7.1/ [24]		
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 62		
Test purpose	SIP-I support. Called party subaddress can be correctly transferred in the		
	Access Transport parameters.		
	BICC/ISUP - SIP-I interworking applies in the originating network User A is		
	located in network A and user B is located in network B. Ensure that an		
	ISUP/BICC ATP parameter present in the encapsulated IAM of the INVITE		
	request and contains a Called party subaddress.		
Configuration	User A is subscribed to the SUB supplementary service		
SIP Parameter	INVITE		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any haundary nama]		
	[any boundary name] Content-Type: application/isup;version=itu-t92		
	Content-Type: application/isup,version=itu-is2 Content-Disposition: signal;handling=required		
	IAM		
	Access transport		
	Called party subaddress		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(IAM)		
	Apply post test routine		
Comments	Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final		
	response?		
	Check: Is an ISUP/BICC ATP parameter present in the encapsulated ANM		
	containing a Called party subaddress? Repeat this test in reverse direction.		
	Inchear mis resum tenerse anechon.		

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		
0	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) 		
	← 200 OK INVITE(ANM) ACK		
	-		
Comments	Apply post test routine Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final		
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.		
	proposition tost 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		
	ISDN Subscriber initiated		
	INFO		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	RES		
	Suspend/resume indicator		
	ISDN subscriber initiated		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	A confirmed session already exists		
	INFO(SUS) →		
	← 200 OK INFO		
	INEO/PEO		
	INFO(RES) → 200 OK INFO		
	=======================================		
Comments	Apply post test routine A session is already established		
Comments	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.		
	Interest the test in reverse unconstitution		

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		
Configuration	containing a Cause value set to #102.		
Configuration SIP Parameter	User A is subscribed to the Terminal Portability supplementary service		
SIP Parameter			
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	SUS		
	Suspend/resume indicator		
	ISDN subscriber initiated		
	102110440011001111110404		
	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) →		
Comments	A session is already established		
Comments	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.		
<u> </u>			

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	INVITE: Request line	
		<pre><<cc> <ndc> <sn>[;npdi][; rn=(Number portability routing number)] </sn></ndc></cc></pre> <pre></pre> <pre><pre></pre><pre></pre><pre></pre><pre></pre><pre></pre><pre><pre></pre><pre></pre><pre></pre><pre></pre><pre></pre><pre><pre></pre><pre></pre><pre></pre><pre></pre><pre></pre><pre><pre></pre><pre></pre><pre></pre><pre></pre><pre></pre><pre></pre></pre></pre></pre></pre>
Message flow		
SIP (Network A)		Interconnection Interface SIP (Network B) INVITE →
Apply post test routine		
Comments	Check:	Is the URI in the userinfo of the Request line in a global number format?
	Check:	Is the URI rn parameter containing the Number Portability Routing
		Number in a global number format?
	Check:	Is optional the URI parameter 'npdi' present?
	Check:	Is the user parameter set to 'phone'?
	Repeat tl	his test in reverse direction.

SS_NP_002		
SIP-SIP/NubP		
5.3, 5.4/ [2]		
NOT [Network A] SE 13		
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.		
INVITE: Request line		
sip: + <cc> <ndc> <sn>@<hostname>;user = phone SIP/2.0</hostname></sn></ndc></cc>		
Interconnection Interface SIP (Network B)		
INVITE ->		
Apply post test routine		
Check: Is the URI in the userinfo of the Request line in a global number format		
without npdi parameter and number portability routing number?		
Check: Is the user parameter set to 'phone'?		
Repeat this test in reverse direction.		

7.3 Accounting

Test case number	SS_acc_001		
Test case group	SIP-SIP/ACCOUNTING		
Reference			
SELECTION EXPRESSION			
Test purpose	Comparison of Charging Data Records > 1 s.		
	Accounting of a confirmed session with a duration > 1 s. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.		
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 180 Ringing 200 OK INVITE ACK →		
	Communication		
	BYE →		
_	← 200 OK BYE		
Comments	 Setup a call from network A to network B. Verify is the session confirmed. Terminate the session after 5 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 callduration callsetuptime (optional) Check the duration indicated in the CDR against the duration in the call 		
	trace. 7. Repeat this test in reverse direction.		
	7. Report the test in reverse direction.		

Test case number	SS_acc_002		
Test case group	SIP-SIP/ACCOUNTING		
Reference	OII ///OCCONTING		
SELECTION EXPRESSION	0 1 (0) 1 D D I		
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	Oil Oil ///OCCONTING		
SELECTION EXPRESSION			
Test purpose	Comparison of Charging Data Records > 15 min.		
	Accounting of a confirmed accoing 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		
Configuration	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 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.		
	I		

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	 ← 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. 7. Repeat this test in reverse direction.		

Test case number	SS_acc_005		
Test case group	SIP-SIP/ACCOUNTING		
	SII -SII /ACCOUNTING		
Reference			
SELECTION EXPRESSION			
Test purpose	Comparison of Charging Data Records more than 30 min.		
	Accounting of a confirmed session with a duration of > 30 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.		
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE → 180 Ringing ← 200 OK INVITE ACK → Communication BYE → 200 OK BYE		
Comments	 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 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	00 000		
	SS_acc_006		
Test case group	SIP-SIP/ACCOUNTING		
Reference			
SELECTION EXPRESSION			
Test purpose	Comparison of Charging Data Records more than 60 min.		
	Accounting of a confirmed session with a duration between 60 min and 120 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.		
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
, ,	INVITE →		
	← 180 Ringing		
	← 200 OK 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 at the earliest 61 min and at the latest 119 min.		
	4. Determine the duration of the session from the trace of the call monitor.		
	5. Compare the following information elements indicated in the CDR's of both		
	networks:		
	calling party number		
	called party number		
	timestamp		
	callduration		
	callsetuptime (optional)		
	6. Check the duration indicated in the CDR against the duration in the call		
	<u> </u>		
	trace.		
	7. 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.		
	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.		
	11. Repeat the test in reverse direction.		

Test case number	SS acc 008		
Test case group	SIP-SIP/ACCOUNTING		
Reference			
SELECTION EXPRESSION			
Test purpose	Comparison of Charging Data Records less than 1 s.		
	Accounting of a confirmed session with duration <1 s. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.		
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface INVITE 180 Ringing 200 OK INVITE ACK Communication BYE 200 OK BYE SIP (Network B) SIP (Network B)		
Comments	 Setup a call from network A to network B. Verify is the session confirmed. Terminate the session after 0,9 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 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 009		
Test case group	SIP-SIP/ACCOUNTING		
Reference			
SELECTION EXPRESSION			
Test purpose	Comparison of Charging Data Records session not confirmed.		
	Accounting of an unsuccessful session in the early dialogue. Verify the duration of the call attempt stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface if applicable.		
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 180 Ringing		
	BYE/CANCEL → 200 OK BYE/CANCEL 487 Request Terminated ACK →		
Comments	 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 0001 C	201
	SS_csel_C	
Test case group	SIP-SIP/C	
Reference	5.7.1.10/ [2	
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15	
Test purpose	User selects an operator 'call-by-call'.	
	User A and user B are located in network A. Ensure that user A is able to call	
	user B and user A is able to select network B as a selected carrier 'call-by-call'.	
Configuration	User in ne	twork A is not presubscribed
SIP Parameter	INVITE: Request line	
	sip: + <cc< th=""><th>C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc></th></cc<>	C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc>
	_	
	INVITE: R	equest line
	sip:	: + <cc> <ndc> <sn>;npdi</sn></ndc></cc>
	·	[;rn= <number number="" portability="" routing="">]@<hostname>;</hostname></number>
		user=phone SIP/2.0
Message flow		
SIP (Network A)		Interconnection Interface SIP (Network B)
		INVITE 1 →
	←	INVITE 2
		Apply post test routine
Comments	Check:	Is the optional '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?
		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 information is available in the Request line
		regarding the end user charging in case of Carrier selection.
	Repeat thi	is test in reverse direction.

Test case number	SS_csel	002
Test case group	SIP-SIP/0	
Reference	5.7.1.10/	
SELECTION EXPRESSION		A] SE14 AND [Network B] SE15
Test purpose	User is presubscribed to operator B.	
l est purpose	Oser is presubscribed to operator b.	
	Hear A ar	nd 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	_	etwork A is presubscribed to network B
SIP Parameter		Request line
		C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc>
	ыр. т чо	05 (17205 (017)[,010-(0011101 12)]@ (1100111011105 0001-p110110 011 72.0
	INVITE: F	Request line
		o: + <cc> <ndc> <sn>;npdi</sn></ndc></cc>
	1	[;rn= <number number="" portability="" routing="">]@<hostname>;</hostname></number>
		user=phone SIP/2.0
Message flow		•
SIP (Network A)		Interconnection Interface SIP (Network B)
		<mark>INVITE</mark> 1 →
	←	
	←	INVITE 1 INVITE 2 Apply post test routine
Comments	← Check:	INVITE 1 INVITE 2 Apply post test routine Is the optional 'cic' tel uri parameter present in the Request URI in the
Comments		INVITE 1 INVITE 2 Apply post test routine Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected
Comments	Check:	INVITE 1 INVITE 2 Apply post test routine Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier?
Comments		INVITE 1 INVITE 2 Apply post test routine Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? Is the 'npdi' parameter present in the Request URI of the INVITE
Comments	Check:	INVITE 1 INVITE 2 Apply post test routine Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE 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?
Comments	Check:	INVITE 1 INVITE 2 Apply post test routine Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE 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
Comments	Check: Check:	INVITE 1 INVITE 2 Apply post test routine Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE 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?
Comments	Check: Check:	INVITE 1 INVITE 2 Apply post test routine Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE 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
Comments	Check: Check: Check: NOTE 1:	INVITE 1 INVITE 2 Apply post test routine Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE 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.
Comments	Check: Check: Check: NOTE 1:	INVITE 1 INVITE 2 Apply post test routine Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE 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. It is possible that further information is available in the Request line
Comments	Check: Check: Check: NOTE 1:	INVITE 1 INVITE 2 Apply post test routine Is the optional 'cic' tel uri parameter present in the Request URI in the INVITE 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.

Test case number	SS_csel_003	
Test case group	SIP-SIP/CS	
Reference	5.7.1.10/ [2]	
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15	
Test purpose	User is presubscribed to an operator unequal to B, and overrides the	
	preselection with call-by-call via operator B.	
	User A and user B are located in network A. User A is preselected to a network	
	unequal to network B. Ensure that user A is able to call user B and user A is able	
	to select network B as a selected carrier 'call-by-call'. The preselected carrier is	
	ignored.	
Configuration	User in network A is not 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>	
	NIN (TE D	
	INVITE: Request line	
	sip: + <cc> <ndc> <sn>;npdi</sn></ndc></cc>	
	[;rn= <number number="" portability="" routing="">]@<hostname>;</hostname></number>	
Message flow	user=phone SIP/2.0	
SIP (Network A)	Interconnection Interface SIP (Network B)	
On (Network A)	INVITE 1	
	← INVITE 2	
	Apply post test routine	
Comments	Check: Is the optional 'cic' tel uri parameter present in the Request URI in the	
	INVITE sent from network A to network B identifying the selected	
	carrier?	
	Check: Is the 'npdi' parameter present in the Request URI of the INVITE	
	request sent from network B to network A?	
	Check: Is optional the 'rn' parameter present in the Request URI of the INVITE	
	request sent from network B to network A?	
	NOTE 1: The 'cic' parameter may be absent according national regulations or	
	national agreements.	
	NOTE 2: It is possible that further information is available in the Request line	
	regarding the end user charging in case of Carrier selection.	
	Repeat this test in reverse direction.	

Test case number	SS_csel_004	
Test case group	SIP-SIP/CS	
Reference	5.7.1.10/ [2]	
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15	
Test purpose	User is presubscribed to operator B, and overrides the preselection with	
	call-by-call via operator B.	
	User A and user B are located in network A. User A is preselected to network B.	
	Ensure that user A is able to call user B and user A is able to select network B	
	as a selected carrier 'call-by-call'. The preselected carrier is ignored.	
Configuration	User in network A is presubscribed to network B	
SIP Parameter	INVITE: Request line	
	sip: + <cc> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</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	
Message flow	· · · · · · · · · · · · · · · · · · ·	
SIP (Network A)	Interconnection Interface SIP (Network B)	
,	INVITE 1 →	
	← INVITE 2	
	Apply post test routine	
Comments	Check: Is the optional '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?	
	request sent from network B to network A? NOTE 1: The 'cic' parameter may be absent according national regulations or	
	NOTE 1: The 'cic' parameter may be absent according national regulations or	
	NOTE 1: The 'cic' parameter may be absent according national regulations or national agreements.	
	NOTE 1: The 'cic' parameter may be absent according national regulations or national agreements.NOTE 2: It is possible that further information is available in the Request line	
	NOTE 1: The 'cic' parameter may be absent according national regulations or national agreements.	

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

7.5 Emergency call

Test case number	SS_ecall_001
Test case group	SIP-SIP/EmC
Reference	5.2.10, 5.7.1.14/ [2]
SELECTION EXPRESSION	
Test purpose	Request line in the INVITE.
	User A attempts to call a PSAP located in network B. Ensure that the Request
	line in the INVITE contains the emergency number and a 'rn' parameter
	containing the PSAP routing number. In addition a location information may be
	present:
	Geolocation header and corresponding PIDF-LO Element
	User-to-User header
	National solution to convey location information
	to make location information available for the PSAP.
Configuration	
SIP Parameter	INVITE: Request line
	sip+ <(emergency number)>[; rn =+<(PSAP routing number)]
	@hostname>;user = phone SIP/2.0
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE ->
0	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 PSAP Routing
	Number?
	Check: Is the user parameter set to 'phone'?
	Check: Is the location information present in the initial INVITE request.
	Geolocation header
	PIDF-LO Element XML 'geopriv' sub element
	Or
	User-to-User header
	Or
	National solution
	Repeat this test in reverse direction.

7.6 SIP Support of Charging

Test case number	SS_sipc_	001
Test case group		SIP_charging
Reference	B.2.3/ [19	
	SE 16	5]
SELECTION EXPRESSION		
Test purpose	User A is in case of information	sful session from user A to user B via network B one single tariff. 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 one single tariff covered in a XML MIME body in a reliable hal or successful final response.
Configuration	ľ	
SIP Parameter	INVITE:	
oir raiametei	Supp 18x or 20 Required Context Context messa crgt ch	orted: 100rel 00 OK dire: 100rel entType: application/vnd.etsi.sci+xml ent-Disposition: render; handling=optional ageType margingControlIndicators margingTariff tariffCurrency currentTariffCurrency currentTariffCurrency currencyFactorScale currencyFactor currencyScale tariffDuration subTariffControl tariffControlIndicators iginationIdentification urrency (optional)
Message flow SIP (Network A)	- 1	
SIF (NELWOIK A)		Interconnection Interface SIP (Network B) INVITE →
CASE A	←	18x(crgt)
CASE A	•	PRACK
	←	200 OK PRACK
	~	200 OK PRACK
0405 0	-	000 OK IN //TE/ - 1)
CASE B	←	200 OK INVITE(crgt)
		Apply post test routine
Comments	Check:	Is the supported header in the initial INVITE set to '100rel'
	Check:	Is the Require header in the response containing the tariff information set to '100rel'?
	Check:	Is the messageType ' crgt ' present in a 1xx provisional or a 200 OK INVITE final response?
	Check:	Is the tariffCurrency element set to 'currentTariffCurrency'?
	Check:	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?
		his test in reverse direction.

Test case number	SS_sipc_	002		
Test case group		SIP_charging		
Reference	B.2.3/ [19			
SELECTION EXPRESSION	SE 16	9]		
		ful session from user A to user B via network B several tariffs in		
Test purpose				
	one sequ	uence.		
	Lloor A is	legated in naturally A and naturally B is recognished for sharping (CDD)		
		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 several tariffs in a sequence covered in a XML MIME body in a		
0 \$1 \$1	reliable p	rovisional or successful final response.		
Configuration	15.0.475			
SIP Parameter	INVITE:			
	Supp	orted: 100rel		
	40 00	0.014		
	18x or 20			
		ire: 100rel		
		entType: application/vnd.etsi.sci+xml		
	Conte	ent-Disposition: render; handling=optional		
		ageType		
	crgt			
		nargingControlIndicators		
	ch	nargingTariff		
		tariffCurrency		
		currentTariffCurrency		
		communicationChargeSequenceCurrency		
		currencyFactorScale		
		currencyFactor		
		currencyScale		
		tariffDuration		
		subTariffControl		
		communicationChargeSequenceCurrency		
		currencyFactorScale		
		currencyFactor		
		currencyScale		
		tariffDuration		
		subTariffControl		
		tariffControlIndicators		
		iginationIdentification		
	CL	ırrency (optional)		
Message flow				
SIP (Network A)		Interconnection Interface SIP (Network B)		
		INVITE →		
CASE A	←	18x(crgt)		
		PRACK →		
	←	200 OK PRACK		
CASE B	←	200 OK INVITE(crgt)		
		Apply post test routine		
Comments	Check:	Is the Supported header in the initial INVITE set to '100rel'?		
	Check:	Is the Require header in the response containing the tariff information		
		set to '100rel'?		
	Check:	Is the messageType 'crgt' present in a 1xx provisional or a 200 OK		
		INVITE final response?		
	Check:	Is the tariffCurrency element set to 'currentTariffCurrency'?		
		Are there more than one communicationCharge		
		SequenceCurrency elements present in the currentTariffCurrency		
		element?		
	Check:	Represents the currencyFactorScale in the communicationCharge		
		SequenceCurrency elements the applicable tariffs?		
	Check:			
	Check:	Is the optional element ' currency ' set to 'EUR' if present?		
		nis test in reverse direction.		
	I topout ii	no toot in 1070100 direction.		

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 SIP (Network A)	Interconnection Interface SIP (Network B)		
CASE A	INVITE → 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?		
	Check: Is the optional element 'currency' set to 'EUR' if present? Repeat this test in reverse direction.		

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

Test case number	SS_sipc	_005		
Test case group		SIP_charging		
Reference	B.2.3/ [1			
SELECTION EXPRESSION	SE 16	-		
Test purpose	Success	sful session from user A to use	er B via network B with a next tariff.	
			rk B is responsible for charging (CDP) sure that the network B sends a tariff	
			tch over time covered in a XML MIME	
		a reliable provisional or successfu		
Configuration				
SIP Parameter	INVITE:			
		orted: 100rel		
	18x or 20	00 OK		
		iire: 100rel		
		entType: application/vnd.etsi.sci-		
		ent-Disposition: render; handling	=optional	
	mess crgt	sageType		
		nargingControlIndicators		
	cl	nargingTariff		
		tariffCurrency		
		currentTariffCurrency	SaguenceCurrency	
		communicationChargeSequenceCurrency currencyFactorScale currencyFactor		
		currencyScale		
		tariffDuration subTariffControl tariffControlIndicators tariffSwitchCurrency nextTariffCurrency communicationChargeSequenceCurrency		
		currencyFactorS		
		currencyFac		
		currencySca tariffDuration	ie	
		subTariffControl		
		tariffControlIndicato	re	
		tariffSwitchOverTime	13	
	O	riginationIdentification		
		urrency (optional)		
Message flow			OID (Naturally D)	
SIP (Network A)		Interconnection Interface INVITE	SIP (Network B) →	
CASE A	←	18x(crgt)	7	
OAGE A	•	PRACK	→	
	←	200 OK PRACK	•	
CASE B	←	200 OK INVITE(crgt) Apply post test routine		
Comments	Check:	Is the supported header in the		
	Check:	Is the Require header in the reset to '100rel'?	sponse containing the tariff information	
	Check:		ent in a 1xx provisional or a 200 OK	
	Check:		ement set to 'nextTariffCurrency'?	
	Check:	Represents the currencyFactor		
	Check:	communicationChargeSequence	ceCurrency element the next tariff? ndicated in the tariffSwitchOverTime	
	J.100K.	element?		
	Check:	Is the optional element 'curren	cy' set to 'EUR' if present?	
		his test in reverse direction.	,	

Test case number	SS_sipc_006
Test case group	SIP-SIP/ SIP_charging
Reference	B.2.3/ [19]
SELECTION EXPRESSION	SE 16
Test purpose	Successful change of a current tariff and next tariff during an active session.
	User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a new
	tariff information with several current tariffs and several next tariffs covered in a XML MIME body in an INFO request.
Configuration	
SIP Parameter	INFO ContentType: application/vnd.etsi.sci+xml
	messageType crgt
	chargingControlIndicators chargingTariff tariffCurrency
	currentTariffCurrency
	communicationChargeSequenceCurrency
	currencyFactorScale
	currencyFactor currencyScale
	tariffDuration
	subTariffControl
	communicationChargeSequenceCurrency
	currencyFactorScale
	currencyFactor
	currencyScale tariffDuration
	subTariffControl
	tariffControlIndicators
	tariffSwitchCurrency
	nextTariffCurrency
	communicationChargeSequenceCurrency
	currencyFactorScale
	currencyFactor
	currencyScale
	tariffDuration
	subTariffControl
	communicationChargeSequenceCurrency
	currencyFactorScale currencyFactor
	currencyScale
	tariffDuration
	subTariffControl
	tariffControlIndicators
	tariffSwitchOverTime
	originationIdentification
Managa flow	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	
	me <mark>ssage</mark> Type	
	aocrg	
	chargingControlIndicators	
	addOnCharge	
	addOnChargeCurrency	
	currencyFactor	
	currencyScale originationIdentification	
	currency (optional)	
Message flow	Currency (optional)	
SIP (Network A)	Interconnection Interface SIP (Network B)	
on (notificity)	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.

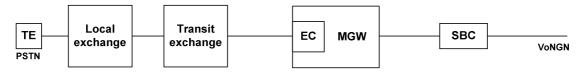


Figure 7.7-2: Reference configuration for PSTN/ISDN with classical access

7.7.1.3 NGN PSTN/ISDN access Configuration

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

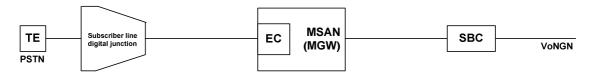


Figure 7.7-3: Reference configuration for NGN with PSTN/ISDN access

7.7.1.4 Access DSL Configuration

Figure 7.7-4 shows the xDSL access configuration.

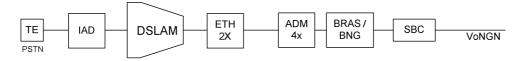


Figure 7.7-4: Reference configuration for DSL access

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

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. D _{tr-seg A-B} = D _{sum-seg A-B} - D _{JB1seg B} D _{tr-seg B-A} = D _{sum-seg B-A} - D _{JB2segA}	
Calling station	The amplitude of the tone is -16 dBm0	
Called station	The amplitude of the tone is -16 dBm0	

Annex A (informative): Bibliography

- IETF RFC 3966 (2004): "The tel URI for Telephone Numbers".
- IETF RFC 3311 (2002): "The Session Initiation Protocol (SIP) UPDATE Method".
- IETF RFC 3323 (2002): "A Privacy Mechanism for the Session Initiation Protocol (SIP)".
- IETF RFC 3325 (2002): "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks".
- ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 Release 10)".
- IETF RFC 2833: "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".
- ETSI TS 134 229-1: "Universal Mobile Telecommunications System (UMTS); Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Part 1: Protocol conformance specification (3GPP TS 34.229-1 version 6.3.0 Release 6)".
- ETSI EG 201 018: "Integrated Services Digital Network (ISDN); Application of the Bearer Capability (BC), High Layer Compatibility (HLC) and Low Layer Compatibility (LLC) information elements by terminals supporting ISDN services".
- ETSI EN 300 093-1: "Integrated Services Digital Network (ISDN); Calling Line Identification Restriction (CLIR) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 207-1: "Integrated Services Digital Network (ISDN); Diversion supplementary services; Digital Subscriber Signalling System No. One (DSS1); Part 1: Protocol specification".
- ETSI EN 300 188-1: "Integrated Services Digital Network (ISDN); Three-Party (3PTY) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 141-1: "Integrated Services Digital Network (ISDN); Call Hold (HOLD) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 185-1: "Integrated Services Digital Network (ISDN); Conference call, add-on (CONF) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 196-1: "Integrated Services Digital Network (ISDN); Generic functional protocol for the support of supplementary services; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 138-1: "Integrated Services Digital Network (ISDN); Closed User Group (CUG) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI TS 124 147: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3 (3GPP TS 24.147 version 9.1.0 Release 9)".
- ETSI EN 300 001: "Attachments to the Public Switched Telephone Network (PSTN); General technical requirements for equipment connected to an analogue subscriber interface in the PSTN".
- ETSI ETS 300 648: "Public Switched Telephone Network (PSTN); Calling Line Identification Presentation (CLIP) supplementary service; Service description".

- ETSI EN 300 092-1: "Integrated Services Digital Network (ISDN); Calling Line Identification Presentation (CLIP) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 659: "Access and Terminals (AT); Analogue access to the Public Switched Telephone Network (PSTN); Subscriber line protocol over the local loop for display (and related) services".
- ETSI TBR 008: "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice; Attachment requirements for handset terminals".
- Recommendation ITU-T Q.951: "Stage 3 description for number identification supplementary services using DSS 1".
- Recommendation ITU-T Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".
- Recommendation ITU-T Q.850 (05/98): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".
- ETSI EG 201 299-1: "Integrated Services Digital Network (ISDN); Network Integration Testing (NIT); ISDN/PSTN end-to-end testing; Part 1: Test Suite Structure and Test Purposes (TSS&TP) specification".

History

Document history			
V1.1.1	August 2012	Publication	
V1.1.2	September 2012	Publication	
V1.2.1	April 2014	Publication	