ETSI TS 101 585 V1.1.2 (2012-09)



IMS Network Testing (INT);
NGN/IMS interconnection tests at the Ic Interface;
Test Suite Structure and Test Purposes (TSS&TP)

Reference

RTS/INT-00084

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

Individual copies of the present document can be downloaded from: http://www.etsi.org

The present document may be made available in more than one electronic version or in print. In any case of existing or perceived difference in contents between such versions, the reference version is the Portable Document Format (PDF). In case of dispute, the reference shall be the printing on ETSI printers of the 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: http://portal.etsi.org/chaircor/ETSI_support.asp

Copyright Notification

No part may be reproduced except as authorized by written permission. The copyright and the foregoing restriction extend to reproduction in all media.

© European Telecommunications Standards Institute 2012. 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	ord	5
1	Scope	6
2	References	6
2.1	Normative references	
2.2	Informative references.	
	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	
7.1.5	Test purposes for Supplementary services	
7.1.5.1	Test purposes for OIP	58
7.1.5.2	Test purposes for OIR	63
7.1.5.3	Test purposes for TIP	66
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		
7.1.5.0 7.1.5.7		
7.1.5.7 7.1.5.8		
7.1.5.8 7.1.5.9	• • • • • • • • • • • • • • • • • • •	
7.1.5.1	1 ' '	
7.1.5.1		
7.1.5.1		
7.1.5.1		
7.1.5.1	4 Completion of Communications to Busy Subscriber (CCBS), Completion of Communication	ns by
716	No Reply (CCNR)	
7.1.6 7.1.6.1	Other PSTN services (SIP-I interworking)	
7.1.6.1 7.1.6.2		
7.1.6.2 7.1.6.3		
7.1.0.3 7.2	Number Portability	
7.3	Accounting	
7.4	Carrier Selection.	
7.5	Emergency call	
7.6	SIP Support of Charging	

7.7	Quality of Service	253
7.7.1	Reference Configurations	253
7.7.1.1	Backbone Configuration	253
7.7.1.2	PSTN/ISDN classic access Configuration	253
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	254
Annex A	(informative): Bibliography	256
History.		258

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 IMS Network Testing (INT).

1 Scope

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

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

2 References

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

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

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

he following refe	erenced 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

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

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

- [9] ETSI TS 124 604: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication Diversion (CDIV) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.604 version 10.3.0 Release 10)".
- [10] ETSI TS 124 605: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Conference (CONF) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.605 Release 10)".
- [11] ETSI TS 124 629: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Explicit Communication Transfer (ECT) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.629 Release 10)".
- [12] ETSI TS 124 611: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Anonymous Communication Rejection (ACR) and Communication Barring (CB) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.611 Release 10)".
- [13] ETSI TS 124 654: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Closed User Group (CUG) using IP Multimedia (IM) Core Network (CN) subsystem, Protocol Specification (3GPP TS 24.654 Release 10)".
- [14] ETSI TS 124 642: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Completion of Communications to Busy Subscriber (CCBS) and Completion of Communications by No Reply (CCNR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification (3GPP TS 24.642 Release 10)".
- [15] ETSI TS 124 615: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification (3GPP TS 24.615 Release 10)".
- [16] ETSI TS 124 606: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Message Waiting Indication (MWI) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.606 Release 10)".
- [17] ETSI TS 124 610: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication HOLD (HOLD) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.610 Release 10)".
- [18] ETSI TS 124 616: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Malicious Communication Identification (MCID) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.616 Release 10)".
- [19] ETSI TS 129 658: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; TISPAN; SIP Transfer of IP Multimedia Service Tariff Information; Protocol specification (3GPP TS 29.658 Release 10)".
- [20] ETSI TS 124 628: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.628 Release 10)".
- [21] IETF RFC 5009 (September 2007): "Private header (P-Header) extension to the Session Initiation Protocol (SIP) for authorization of Early Media".
- [22] ITU-T Recommendation V.152 (November 2004): "Procedures for supporting Voice-Band Data over IP Networks".
- [23] ITU-T Recommendation T.38 (September 2010, prepublished): "Procedures for real-time Group 3 facsimile communication over IP networks".

- [24] ITU-T Recommendation Q.1912.5: "SERIES Q: SWITCHING AND SIGNALLING Specifications of signalling related to Bearer Independent Call Control (BICC) Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".
- [25] ETSI TS 183 036: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ISDN/SIP interworking; Protocol specification".

2.2 Informative references

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

- [i.1] ETSI EN 300 403-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1) protocol; Signalling network layer for circuit-mode basic call control; Part 1: Protocol specification [ITU-T Recommendation Q.931 (1993), modified]".
- [i.2] ISO/IEC 9646 (1994): "Information technology -- Open Systems Interconnection -- Conformance testing methodology and framework".
- [i.3] ETSI TR 102 775 (V1.5.1): "Speech and multimedia Transmission Quality (STQ); Guidance on objectives for Quality related Parameters at VoIP Segment-Connection Points; A support to NGN transmission planners".
- [i.4] ITU-T Recommendation Q.1902.2 (07/2001): "Bearer Independent Call Control protocol (Capability Set 2) and Signalling System No.7 ISDN User Part: General functions of messages and parameters".

3 Definitions and abbreviations

3.1 Definitions

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

BICC or ISUP specific terminology, references to ITU-T Recommendation 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 ISUPexchange) that terminates incoming calls using SIP and originates outgoing calls using the BICC or ISUP protocols

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

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

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

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

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

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

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

NOTE: as defined in RFC 3312 [6].

syntactically invalid: specifies a test purpose covering a signalling procedure where a valid (expected in the current status of the IUT) but not correctly encoded (unknown or incorrect parameter values) message is sent to the IUT, which 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 capitalized.

EXAMPLE 1: INVITE, INFO.

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

EXAMPLE 2: To, From, Call-ID.

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

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

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

EXAMPLE 4: sip: URI, tel: URI.

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

EXAMPLE 5: 100 Trying, 484 Address Incomplete.

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

EXAMPLE 6: 200 OK INVITE, 200 OK PRACK.

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

EXAMPLE 7: m=line, a=line.

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

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

3.2 Abbreviations

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

ACR Anonymous Communication Rejection

CB Communication Barring

CFB Communication Forwarding Busy

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

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

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

CONF Conference

CUG Closed User Group CW Communication Waiting

ECT Explicit Communication Transfer

GW GateWay

HOLD Communication Hold

ISDN Integrated Services Digital Network

IUT Implementation Under Test

MCID Malicious Communication Identification

MWI Message Waiting Indication

OIP Originating Identification Presentation

OIR Originating Identification presentation Restriction

PASP Public Answering Safety Point

PICS Protocol Implementation Conformance Statement

PSTN Public Switched Telephone Network

QoS Quality of service

SIP Session Initiation Protocol

TIP Terminating Identification Presentation
TIR Terminating Identification Restriction

TP Test Purpose
TSS Test Suite Structure

4 Test Suite Structure (TSS)

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

5 Declarations

5.1 Numbering Scheme

FFS

5.2 Reference configuration

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

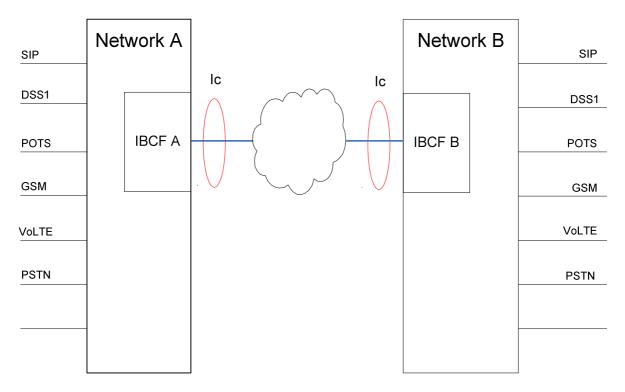


Figure 5.2-1: Reference configuration for the interconnection test

5.3 Selection of End Devices

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

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

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

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

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

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

6 Selection Expressions

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

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

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

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

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

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

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

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

SELECTION EXPRESSION:		Support	Support
		Network A	Network B
SE 29:	The Network supports Communication Deflection?		
SE 30:	The Network supports the CDIV Notification procedure?		
SE 31:	The Network supports conference (CONF)		
SE 32:	The Network supports the Communication Barring procedure (CB) -		
	(Black list for incoming calls)?		
SE 33:	The Network supports the Anonymous Communication Rejection		
	(ACR)?		
SE 34:	The Network supports the Closed User Group (CUG)?		
SE 35:	The Network supports the Communication Waiting (CW) service?		
SE 36:	The Network supports the T _{AS-CW} timer?		
SE 37:	The Network supports Explicit Communication Transfer (ECT)?		
SE 38:	The network supports Malicious Communication Identification (MCID)		
SE 39:	The Network supports Message Waiting Indication (MWI)?		
SE 40:	The Network supports Completion of Communications to Busy		
	Subscriber (CCBS)?		
SE 41:	The Network supports Completion of Communications by No Reply		
	(CCNR)		
OF 40	Terminal capabilities		1
SE 42:	The End device (in Network B) establishes an Early dialogue by		
	sending a 183 AND The Network B allows the bearer transmission in		
SE 43:	the early dialogue 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 wilder T.38 codec		
SE 46:	A SIP end device is used supporting a ISDN user equipment and the		
OL 40.	PSTN XML Schema is used		
SE 47:	End device is located in the PSTN or PLMN		
SE 48:	The terminating UE supports the from-change tag procedure and		
	sends a second user identity in an UPDATE request after the		
	dialogue is confirmed		
SE 49:	The end device performs ECT using the 'Blind/assured transfer'		
SE 50:	The end device performs ECT using the 'Consultative transfer'		
SE 51:	The end device supports the Resource reservation procedure		
	PSTN/PLMN Supplementary services	-	
SE 52:	CLIP/CLIR is supported in the PSTN/PLMN part of the network		
SE 53:	COLP/COLR is supported in the PSTN/PLMN part of the network		
SE 54:	HOLD is supported in the PSTN/PLMN part of the network		
SE 55:	CDIV is supported in the PSTN/PLMN part of the network		
SE 56:	CONF/3PTY is supported in the PSTN/PLMN part of the network		
SE 57:	ACR is supported in the PSTN/PLMN part of the network		
SE 58:	CUG is supported in the PSTN/PLMN part of the network		
SE 59:	CW is supported in the PSTN/PLMN part of the network		
SE 60:	ECT is supported in the PSTN/PLMN part of the network		
SE 61:	MCID is supported in the PSTN/PLMN part of the network		
SE 62:	SUB is supported in the PSTN/PLMN part of the network		
SE 63:	UUS is supported in the PSTN/PLMN part of the network		
SE 64:	TP is supported in the PSTN/PLMN part of the network		

7 Test purposes

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

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

7.1 Testing of SIP protocol requirements

7.1.1 Test purposes for Basic call, Successful

Test case number	SS_bcall_001
Test case group	BCALL/successful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	Basic call normal call clearing from the called user.
	Ensure that call establishment is performed correctly. In the active call state ensure the property of speech. The call is released from the called user.
Configuration	
SIP Parameter	
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE + 100 Trying + 180 Ringing + 200 OK INVITE ACK Communication BYE 200 OK BYE →
Comments	Establish a communication from network A to Network B Check: Ensure the property of speech. Check: Are the media streams terminated after the 200 OK BYE was sent? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

<u> </u>	
Test case number	SS_bcall_002
Test case group	BCALL/successful
Reference	[4]
SELECTION EXPRESSION	
Test purpose	Basic call normal call clearing from the calling user.
-	
	Ensure that call establishment is performed correctly. In the active call state
	ensure the property of speech. The call is released from the calling user.
Configuration	
SIP Parameter	
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
,	INVITE →
	← 100 Trying
	← 180 Ringing
	← 200 OK INVITE
	ACK →
	Communication
	BYE →
	← 200 OK BYE
Comments	Establish a communication from network A to Network B
	Check: Ensure the property of speech.
	Check: Are the media streams terminated after the 200 OK BYE was sent?
	Repeat this test in reverse direction.
	Repeat this test with all chosen end devices.

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

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

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

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

Test case number	SS_bcall_007	
Test case group	BCALL/successful	
Reference	8/[21]	
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE3 AND SE 42	
Test purpose	P-Early-Media header supported early dialogue with 183.	
	Ensure that an early dialogue is established by sending a 183 Session Progress from Network B and the P-Early-Media header is present authorizes early media.	
Configuration		
SIP Parameter	INVITE P-Early-Media: supported SDP	
	183 P-Early-Media: [any value authorizes early media] SDP	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 183 Session Progress Apply post test routine	
Comments	Establish a communication from network A to Network B Check: Is a 183 sent from Network B to establish an early dialogue? Check: A bearer transmission is possible in backward directions. Repeat this test in reverse direction.	

SS_bcall_008	
BCALL/successful	
8/[21]	
[Network A] SE 3 AND [Network B] SE 3	
P-Early-Media header supported early dialogue with 180.	
Ensure that an early dialogue is established by sending a 180 Ringing from	
Network B and the P-Early-Media header is present authorizes early media.	
, , ,	
INVITE	
P-Early-Media: supported	
SDP	
180	
P-Early-Media: [any value authorizes early media]	
SDP	
JUF	
lateres and a three lateres are CID (Naturals D)	
Interconnection Interface SIP (Network B)	
INVITE →	
← 180 Ringing	
Apply post test routine	
Establish a communication from network A to Network B	
Check: Is a 183 sent from Network B to establish an early dialogue?	
Check: A bearer transmission is possible in backward directions.	
Repeat this test in reverse direction.	

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

Test case number	SS_bcall_010		
Test case group	BCALL/successful		
Reference	8/[21]		
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE 3 AND SE 35		
Test purpose	P-Early-Media header supported early dialogue with 182.		
1000 pan pood	Lany media media emperior carry dialogue mai lozi		
	Ensure that an early dialogue is established by sending a 182 Queued from		
	Network B and the P-Early-Media header is present authorizes early media. The		
	Call is a waiting call in network B.		
Configuration			
SIP Parameter	INVITE		
	P-Early-Media: supported		
	SDP		
	182		
	P-Early-Media: [any value authorizes early media]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE →		
	← 180 Call Is Being Forwarded		
	Apply post test routine		
Comments	Establish a communication from network A to Network B		
	Check: Is a 181 sent from Network B to establish an early dialogue?		
	Repeat this test in reverse direction.		

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

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

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

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

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

Test case number	SS_bcall_016		
Test case group	BCALL/successful		
Reference	5.10/[2]		
SELECTION EXPRESSION			
Test purpose	Route header in the ACK.		
	Ensure that the Route header is present in ACK from network A upon connection establishment from network A is completed and the tops header or entry is set to the IBCF of network B.		
Configuration			
SIP Parameter	ACK:		
	Route: <address b="" ibcf="" in="" network="" of="">;lr,</address>		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network	(B)	
	INVITE →		
	€ 180 Ringing		
	€ 200 OK INVITE		
	ACK →		
	Apply post test routine		
Comments	Comments Establish a communication from network A to Network B Check: Route header is present in the ACK and the topmost head		
	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.	
Configuration	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>	
SIP Parameter	INVITE:	
	Content-Type: application/sdp m=audio <port number=""> RTP/AVP TYPE_SDP= PIXIT (table 7.1.1-1) or m= Image <port number=""> Udptl or Tcptl TYPE_SDP= PIXIT (table 7.1.1-1) a=TYPE_SDP= PIXIT (table 1) b=TYPE_SDP= PIXIT (table 1)</port></port>	
Message flow	Interconnection Interfere	
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.	
	Inepeat this test with all thosen end devices.	

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

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

Table 7.1.1-1

TYPE	E_SDP	m= line		b= line	a= line
VA	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwidth-value></bandwidth-value></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding</clock></encoding></dynamic-pt>
				(see note)	parameters>
VA_01	Audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A or rtpmap 0 PCMU/8000
VA_02	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> PCMU/8000</dynamic-pt>
VA_03	Audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A or rtpmap 8 PCMA/8000
VA_04	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> PCMA/8000</dynamic-pt>
VA_05	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> CLEARMODE</dynamic-pt>
NOTE:	NOTE: <bandwidth value=""> for <modifier> of AS is evaluated to be B kbit/s.</modifier></bandwidth>				

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

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

Test case number	SS bcall 022		
Test case group	BCALL/successful		
Reference	[5] and [23]		
SELECTION EXPRESSION	[Network A] SE 45 AND [Network B] SE 45		
Test purpose	Fax transmission using the T.38 in an audio m-line codec.		
	Ensure that a Fax transmission is possible from Network A to Network B and the		
	relevant codec is the T.38 in an 'audio' m-line codec. Ensure in the active call		
	state the property of Fax transmission.		
Configuration			
SIP Parameter	INVITE: SDP		
	m=image <port> udptl t38</port>		
	180/200 OK INVITE: SDP		
	m=image <port> udptl t38</port>		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
,	INVITE (SDP1)		
	← 180 Ringing		
	← 200 OK INVITE (SDP2)		
	ACK →		
	Apply post test routine		
Comments	Establish a communication from network A to Network B		
	Check: Contains the SDP offer in the initial INVITE a T.38 codec in an 'audio'		
	line.		
	Check: Contains the SDP answer in the 180 or 200 OK INVITE a T.38 codec		
	in an 'audio' line.		
	Check: Is Fax transmission is successful?		
	Repeat this test in reverse direction.		

Test case number	SS_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			
Test purpose	Overla	Overlap sending, the Multiple INVITE method is used.		
		that call establishment using overlap		
		that in the confirmed state the voice	transfer or	n the media and B-channels
	is perfo	rmed correctly.		
Configuration				
SIP Parameter				
Message flow				41
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 Check:	sh a communication from ISDN to SIF		
	SID one			
		swers with 180 Ringing. this test in reverse direction.		
L	Inchear	una teat in reverse unection.		

Test case number	SS_bcall_024					
Test case group	BCALL/successful					
Reference	4.9, N/[2]					
SELECTION EXPRESSION	[Network A] SE 47 AND [Network A] SE 4 AND [Network B] SE 5					
Test purpose	Overlap sending, the in-Dialogue method is used					
	Ensure that call establishment using overlap sending is performed correctly. Ensure that in the confirmed state the voice transfer on the media and B-channels is performed correctly.					
Configuration						
SIP Parameter	INVITE 2: Supported: 100rel					
	183: Require: 100rel					
	INFO: Content-Type: application/x-session-ii SubsequentDigit: <additional digits=""></additional>	nfo				
Message flow	1 0					
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	SIP (Network B) → → → →				
	← 200 OK INFO← 180 Ringing(CSq 2)					
Comments	Apply post test routine	CID using the quarter analytics in ICDN				
Comments	Establish a communication from ISDN to Check: All INVITE requests contains the Check: The 183 session Progress that Require header set to 100rel. Check: All INFO requests contain the C session-info'. Check: All INFO requests contains the the additional digits. The UE B answers with 180 Ringing resp	e same Call ID and From header values. establishes an early dialogue contains a content-Type header set to 'application/x-'SubsequentDigit:' MIME body containing				

Test case number	SS_bcall_025		
Test case group	BCALL/succ	essful	
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		ISDN access either in the PSTN or the SIP - ISDN interworking	
	according [10	D] applies	
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional		
	<pre><?xml version="1.0" encoding="utf-8"?> PSTN BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>ITC_value</pre> <pre>< BCoctet4 TransferMode>00< InformationTransferRate>10000</pre> BCoctet5 Layer1Identification>01 UserInfoLayer1Protocol>00011		
Message flow SIP (Network A)		terconnection Interface SIP (Network B) INVITE Apply post test routine	
Comments		a PSTN XML MIME body contained in the INVITE request?	
	Check: Is	the BearerCapability element is present?	
	Check: Is InformationTransferCabability element is set as indicated in table 2.1.1-1? Check: Is the InformationTransferCabability element value consistent with the codec list in the SDP? Check: Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP? Repeat this test in reverse direction.		
<u>L</u>	I topout and t	occurrence amountm	

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		
Manageraflaw	HighLayerCharacteristics>[any value]<		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)		
SIF (Network A)	Interconnection Interface SIP (Network B) INVITE →		
	taran da antara da a		
Comments	Apply post test routine Check: Is a PSTN XML MIME body contained in the INVITE request?		
Comments	Check: Is the HighLayerCapability element is present?		
	Repeat this test in reverse direction.		
	propositions test in reverse unection.		

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				
	Overly repairs 114 Off an adding 11944 Off.				
	<pre><?xml version="1.0" encoding="utf-8"?> PSTN</pre>				
	ProgressIndicator				
	ProgressOctet3 CodingStandard>00<				
	Location>yyyy<				
	ProgressOctet4				
	ProgressDescription>0000110<				
	ProgressIndicator				
	ProgressOctet3				
	CodingStandard>00<				
	Location>0000<				
	ProgressOctet4				
	ProgressDescription>[any value]<				
Message flow					
SIP (Network A)	Interconnection Interface SIP (Network B)				
	INVITE →				
	Apply post test routine				
Comments	Check: Is a PSTN XML MIME body contained in the INVITE request?				
	Check: Is a ProgressIndicator element present and the ProgressDescription				
	element is set to '0000110'? Check: Is optional a second ProgressIndicator element present and the				
	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.				
	Inspeat this test in reverse uncetion.				

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 varsion="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					
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				
	Progress Description element is set to any value not #2 and not #8?				
	Repeat this test in reverse direction.				

Test case number	SS healt 020			
	SS_bcall_029			
Test case group	BCALL/successful			
Reference	5.1.2.3/[25]			
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6			
Test purpose	PSTN XML ProgressIndicator element in the 200.			
	User B is located in network B and an ISDN end device is used. Ensure that the			
	200 OK INVITE response contains a PSTN XML MIME body and at least one			
	ProgressIndicator element is present.			
Configuration	User B is an ISDN access either in the PSTN or the SIP - ISDN interworking			
	according [10] applies			
SIP Parameter	200:			
	Content-Type: application/vnd.etsi.pstn+xml			
	Content-Disposition: signal;handling=optional			
	xml version="1.0" encoding="utf-8"?			
	PSTN			
	ProgressIndicator			
	ProgressOctet3			
	CodingStandard>00<			
	Location>yyyy<			
	ProgressOctet4			
	ProgressDescription>0000111<			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE >			
	← 180 Ringing			
	← 200 OK INVITE			
	ACK →			
	Apply post test routine			
Comments	Check: Is a PSTN XML MIME body contained in the 200 OK INVITE			
	response?			
	Check: Is a ProgressIndicator element present and the ProgressDescription			
	element is set to '0000110'?			
	Repeat this test in reverse direction.			

Test case number	SS_bcall_030				
Test case group	BCALL/successful				
Reference	5.1.1.2/[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 "1.0" enceding "uff 9"2				
	xml version="1.0" encoding="utf-8"?				
	PSTN RegrerCapability				
	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	Tel. 1. 2000 Marie III de la companya del companya della companya				
Comments	Check: Is a PSTN XML MIME body contained in the INVITE request?				
	Check: Is the first BearerCapability InformationTransferCabability element is				
	set as indicated to '00000'? Check: Is the second BearerCapability InformationTransferCabability element				
	is set as indicated to '10001'?				
	Check: Is the InformationTransferCabability element value consistent with the				
	codec list in the SDP?				
	Check: Is the InformationTransferCabability element value consistent with the				
	bandwidth information in the SDP?				
	Repeat this test in reverse direction.				
	1 1				

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

Test case number	SS_bcall	_032			
Test case group	BCALL/s	uccessful			
Reference	5.1.2.3/[25]				
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6				
Test purpose	Fall back occurs.				
1 3 5 5 Fair P 3 5 5		. • • • • • • • • • • • • • • • • • • •			
	User B is	located in network B and an ISDN end device is used. The Fallback			
		on type was requested in the initial INVITE request. Ensure that the 200			
	OK INVIT	TE response contains a PSTN XML MIME body and a BearerCapability			
	element is present the InformationTransferCabability element set to '00000'. A				
	PSTN XI	PSTN XML MIME ProgressIndicator body is present, the ProgressDescription is			
		set to '0000101'.			
Configuration	User B is an ISDN access either in the PSTN or the SIP - ISDN interworking				
_	according	g [10] applies			
SIP Parameter	200: Content-Type: application/vnd.etsi.pstn+xml				
	Content-Disposition: signal;handling=optional				
	xml version="1.0" encoding="utf-8"?				
	PSTN				
		erCapability			
	B	Coctet3			
	CodingStandard>00<				
	InformationTransferCabability>00000<				
	ProgressIndicator ProgressOctet4				
	"	ProgressDescription>0000101<			
Message flow		1 TogressDescription>0000 ToTC			
SIP (Network A)		Interconnection Interface SIP (Network B)			
on (notwork /t)		INVITE →			
	•	180 Ringing			
	•	200 OK INVITE			
		ACK →			
		Apply post test routine			
Comments	Check:	Is a PSTN XML MIME body contained in the 200 OK INVITE			
		response?			
	Check:	Is a BearerCapability element present, the			
		InformationTransferCabability element set to '00000'?			
	Check:	Is a ProgressIndicator element is present, the ProgressDescription is			
		set to '0000101'?			
	Check:	Is the InformationTransferCabability element value consistent with the			
	 	codec list in the SDP?			
	Check:	Is the InformationTransferCabability element value consistent with the			
		bandwidth information in the SDP?			
	Repeat th	nis test in reverse direction.			

Test case number	SS_bcall_033			
Test case group	BCALL/successful			
Reference	7.1/[24]			
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47			
Test purpose	SIP-I support, Basic call, IAM present in the INVITE request.			
rest purpose	on Tsupport, Busic buil, IAM present in the INVITE request.			
	Ensure that when a call initiated in the PSTN or the PLMN and the ISUP -			
	SIP-I interworking is applicable in the originating network, a ISUP IAM is			
	encapsulated in the initial INVITE request.			
	Ensure that all the mandatory parameters in the IAM are present and the values			
	are valid and the Transmission medium requirement parameter is consistent			
	with the SDP.			
Configuration				
SIP Parameter	INVITE:			
	Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name]			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	1444			
	IAM			
	Nature of connection indicators			
	Forward call indicators			
	Calling party's category			
	Transmission medium requirement Called party number Calling party number (optional) Optional forward call indicators (optional)			
	Hop counter (optional) User service information (optional) Access transport (optional)			
	Access transport (optional)			
	[any boundary name]			
Message flow	[arry boardary riamo]			
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.				
Configuration	Ensure that the original IAM is encapsulated in any INVITE request.				
Configuration SIP Parameter					
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)				
SIF (Network A)	INVITE(1)				
	← 484 Address Incomplete(1)				
	ACK →				
	INVITE(2)				
	← 484 Address Incomplete(2)				
	ACK →				
	INVITE(3) →				
	← 484 Address Incomplete(3)				
	ACK →				
	•				
	INVITE(4) →				
	€ 180 Ringing(4)				
	Apply post test routine				
Comments	Establish a communication from network A to Network B using the overlap				
	procedure in Network A				
	Check: Are the INVITE requests sent with the same From tag and the Call-ID?				
	Check: After the 180 applies, are all previous INVITE transactions are				
	terminated with a 484 final response?				
	Check: Is the encapsulated IAM present in the initial INVITE request also				
	encapsulated in any following INVITE request required for the call				
	setup?				
	Repeat this test in reverse direction.				

Test case number	SS_bcall_035					
Test case group	BCALL/successful					
Reference	6.5/[24]					
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47					
Test purpose	SIP-I support, Basic call, ACM present in the 180 response.					
	Ensure that on receipt of a 180 Ringing provisional response and an					
	SIP-I - ISUP interworking is applicable in the terminating network the Backward					
	call indicators parameter in the encapsulated ACM is present and the value 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]					
	[and have day and a					
	[any boundary name]					
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required					
	Content-Disposition. Signal, nanding=required					
	ACM					
	Backward call indicators					
	[any boundary name]					
Message flow						
SIP (Network A)	Interconnection Interface SIP (Network B)					
	INVITE →					
	← 100 Trying					
	← 180 Ringing(ACM)					
	Apply post test routine					
Comments	Establish a communication from network A to Network B					
	Check: Is an ISUP ACM message encapsulated in the 180 Ringing provisional					
	response? Check: Is the mandatory Backward call indicators parameter present in the					
	encapsulated ISUP ACM and are the values valid?					
	Check: Are the values of optional parameters in the encapsulated ISUP ACM					
	valid?					
	Check: If an SDP answer is present in the 180, are the codec and the					
	bandwidth information in the 'a' attributes consistent with Transmission					
	medium requirement in the encapsulated IAM of the INVITE request?					
	Repeat this test in reverse direction.					
	1 -1					

Test case number	SS_bcall_036					
Test case group	BCALL/successful					
Reference	6.5/[24]					
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47					
Test purpose	SIP-I support. Basic call, early ACM present in the 183 response.					
	Ensure that on receipt of a 183 Session Progress provisional response and an SIP-I - ISUP interworking is applicable in the terminating network the Backward call indicators parameter in the encapsulated ACM is present and the value of the Called party's status indicator is set to 'no indication'. Ensure that the values of the optional parameters in the encapsulated ACM are valid.					
Configuration	Select a proper destination that sends an early ACM in the PSTN/PLMN e.g. announcement					
SIP Parameter	183: Content-Type: multipart/mixed;boundary=[any boundary name]					
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicators Called party's status indicator= no indication					
Manageraflau	[any boundary name]					
Message flow 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						
	LVGIL IIIUICALOI = ALENTING						
	[any boundary name]						
Message flow	[any boundary hamo]						
SIP (Network A)	Interconnection Interface SIP (Network B)						
,	INVITE -						
	← 100 Trying						
	€ 183 Session Progress(ACM)						
	← 180 Ringing(CPG)						
	Apply post test routine						
Comments	Establish a communication from network A to Network B						
	Check: Is an ISUP CPG message encapsulated in the 180 Ringing provisional						
	response?						
	Check: Is the mandatory Event indicator present in the encapsulated ISUP						
	CPG set to 'ALERTING'?						
	Check: Are the values of optional parameters in the encapsulated ISUP CPG						
	valid?						
	Repeat this test in reverse direction.						

Test case number	SS_bcall_038					
Test case group	BCALL/successful					
Reference	6.7/[24]					
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47					
Test purpose	SIP-I support. Basic call, ANM present in a 200 OK INVITE response.					
	Ensure that on receipt of a 200 OK INVITE final response and an SIP-I - ISUP interworking is applicable in the terminating network the ISUP ANM is encapsulated in the 200 OK. Ensure that the values of the optional parameters in the encapsulated ANM are					
	valid.					
Configuration						
SIP Parameter	180: Content-Type: multipart/mixed;boundary=[any boundary name]					
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM					
	[any boundary name]					
Message flow	[any boundary name]					
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 th 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						
rest purpose	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) →						
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?						
	Check: Is the ISUP RLC encapsulated in the 200 OK BYE? Repeat this test in reverse direction.						

7.1.2 Codec negotiation

Tables and an annual and						
Test case number	SS_codec_001					
Test case group	BCALL/Codec_Negotiation					
Reference	[3], [4] and [5]					
SELECTION EXPRESSION						
Test purpose	Session update requested by the calling user.					
	During the session, the calling user decides to change the characteristics of the					
	media session. This is accomplished by sending a re-INVITE or UPDATE					
	containing a new media description. This re-INVITE or UPDATE references the					
	existing dialog so that the other party knows that it is to modify an existing					
	session instead of establishing a new session. The other party sends a 200 (OK)					
	to accept the change. The requestor responds to the 200 (OK) with an ACK.					
	In case when the parameter in the SDP rtpmap: <dynamic-pt> is used the</dynamic-pt>					
Configuration	codecs in table 7.1.2-1 applies.					
Configuration SIP Parameter	0000					
SIP Parameter	SDP1: codec x chosen from table 7.1.2-1 SDP3: codec y chosen from table 7.1.2-1					
Managara flavo	SUP3: codec y chosen from table 7.1.2-1					
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)					
SIF (Network A)	A confirmed session already exists (SDP 1)					
CASE A	INVITE(SDP3)					
ONOL N	€ 200 OK INVITE(SDP4)					
	ACK →					
CASE B	UPDATE(SDP3) →					
	← 200 OK UPDATE(SDP4)					
	Apply post test routine					
Comments	Establish a communication from network A to Network B using SDP1 chosen					
	from the table 7.1.2-1					
	Check: The calling user changes the media description using INVITE request					
	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: 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					
	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.					

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	·					
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)					
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					
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 massage.					
	OK INVITE message. Ensure that in the confirmed call state the voice transfer on the media channels					
	is performed correctly.					
Configuration	no perioritied correctly.					
SIP Parameter	INVITE: SDP offer					
	200: SDP answer					
Message flow						
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(SDP1) ★ 180 Ringing					
200 OK INVITE(SDP2) ACK →						
						Apply post test routine
Comments	Establish a communication from network A to Network B					
Check: Is the SDP offer contained in the initial INVITE request?						
	Check: Is the SDP answer contained in the 200 OK INVITE final response?					
	Repeat this test in reverse direction.					

Table: 7.1.2-1

VARIABLE	PT	Encoding	media	clock	channels	Supported in	Supported in
			type	rate		network A	network B
VA_01	0	PCMU	Α	8,000	1		
VA_02	3	GSM	Α	8,000	1		
VA_03	4	G723	Α	8,000	1		
VA_04	5	DVI4	Α	8,000	1		
VA_05	6	DVI4	А	16,000	1		
VA_06	7	LPC	А	8,000	1		
VA_07	8	PCMA	А	8,000	1		
VA_08	9	G722	А	8,000	1		
VA_09	10	L16	Α	44,100	2		
VA_10	11	L16	А	44,100	1		
VA_13	12	QCELP	А	8,000	1		
VA_12	13	CN	Α	8,000	1		
VA_13	14	MPA	А	90,000			
VA_14	15	G728	Α	1 8,000	1		
VA_15	16	DVI4	Α	11,025	1		
VA_16	17	DVI4	Α	22,050	1		
VA_17	18	G729	Α	8,000	1		
VA_18	Dyn	G726-40	Α	8,000	1		
VA_19	Dyn	G726-32	Α	8,000	1		
VA_20	Dyn	G726-24	Α	8,000	1		
VA_21	Dyn	G726-16	Α	8,000	1		
VA_22	Dyn	G729D	Α	8,000	1		
VA 23	Dyn	G729E	Α	8,000	1		
VA_24	Dyn	GSM-EFR	Α	8,000	1		
VA_25	25	CelB	V	90,000			
VA_26	26	JPEG	V	90,000			
VA_27	28	Nv	V	90,000			
VA_28	31	H261	V	90,000			
VA_29	32	MPV	V	90,000			
VA_30	33	MP2T	V	90,000			
VA_31	34	H263	V	90,000			
VA_32	Dyn	H263-1998	V	90,000			
VA_33	Dyn	AMR	Α	8,000	1		
VA_34	Dyn	AMR-WB	Α	16,000	1		
VA_35	Dyn	telephone- event	А	8000	1		

47

7.1.3 Resource Reservation

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

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

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

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

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

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

Test case number	SS_unsucc_005	
Test case group	BCALL/unsuccessful	
Reference	[4]	
SELECTION EXPRESSION		
Test purpose	The called user is not available under	er the called number.
	Ensure that when the number is change	ed, the network initiate call clearing
	to the calling user with a 410 Gone me	ssage.
Configuration		
SIP Parameter		
Message flow		
SIP (Network A)	Interconnection Interface	SIP (Network B)
	INVITE →	
	← 410 Gone	
	ACK →	
Comments	Establish a communication from network A to Network B, user B is not	
	allocated in Network B.	
	Check: Is a 410 Gone sent from Ne	twork 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 undianged.	
SIP Parameter	INIVITE, and a material in Naturals D	
Message flow	INVITE: codec not supported in Network B	
SIP (Network A)	Interconnection Interface SIP (Network B)	
SIF (NetWORK A)	INVITE ->	
	← 180 Ringing	
	€ 200 OK INVITE	
	ACK →	
	Communication	
	INVITE →	
	← 488 Not Acceptable Here	
	ACK →	
	Apply post test routine	
Comments	Establish a communication from network A to Network B.	
	User A in Network A attempts to change the session by sending a SDP offer to	
	the UE in Network B.	
	Network B does not support the codec sent in the offer.	
	Check: Is a 488 Not Acceptable Here sent from Network B to Network A?	
	Repeat this test in reverse direction.	

Test case number	SS_unsucc_008	
Test case group	BCALL/unsuccessful	
Reference	[3], [4] and [5]	
SELECTION EXPRESSION		
Test purpose	Session update requested by the called user is unsuccessful, existing session remains unchanged.	
	During the session, the called user decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. Ensure that if the other party does not accept the change, he sends an error response such as 488 Not Acceptable Here, which also receives an ACK. The session remains unchanged.	
	The 488 Not Acceptable Here may be sent by a simulation equipment.	
Configuration		
SIP Parameter	INVITE: codec not supported in Network A	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 180 Ringing COUNTY OF THE NAME OF T	
	CK →	
	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_unsuc	cc_009		
Test case group	BCALL/un	successful		
Reference	[4]			
SELECTION EXPRESSION				
Test purpose	Call clear user.	Call clearing due to no answer from the called user initiated by the calling user.		
		at when there is no answer fro all clearing to the called user w		,
Configuration				
SIP Parameter				
Message flow SIP (Network A)		Interconnection Interface	_	SIP (Network B)
	←	INVITE 180 Ringing CANCEL/BYE	→	
	←	200 OK CANCEL/BYE		
	(487 Request Terminated ACK	→	
Comments	Check: Is a CANCEL or BYE request is sent from the originating user? Check: Is a 487 Request Terminating send from the terminating user? Check: Are the media streams terminated after the 200 OK CANCEL/BYE was sent?			
	Repeat thi	is 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	
	luser.	
	Ensure that, when the called user does not accept the Media session, the called	
	user initiate call clearing to the calling user with 488 Not Acceptable Here, which	
	also receives an ACK.	
Configuration		
SIP Parameter	INVITE: codec not supported at user (Network B)	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
	→ INVITE → 488 Not Acceptable Here ←	
	· ·	
	→ ACK →	
Comments	Establish a call setup from network A to Network B.	
	User B in Network B does not support the codec offered in the SDP received	
	from Network A.	
	Check: Is a 488 Not Acceptable Here sent from Network B to Network A.	
	Repeat this test in reverse direction.	

Test case number	ISS UNDUGG 044	
	SS_unsucc_011	
Test case group	BCALL/unsuccessful	
Reference	[4]	
SELECTION EXPRESSION		
Test purpose	Call clearing due to no answer from the called user initiated by the originating network.	
	Ensure that when there is no answer from the called user, the originating	
	network initiate the call clearing after timeout of SIP timer C and sends a	
	CANCEL or BYE to the called user.	
Configuration		
SIP Parameter		
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
,	→ INVITE →	
	← 180 Ringing	
	Start timer C	
	Timeout timer C	
	CANCEL/BYE →	
	€ 200 OK CANCEL/BYE	
	← 487 Request Terminated	
	ACK →	
Comments	Check: Is a CANCEL or BYE request is sent by the originating network?	
	Check: Is a 487 Request Terminating send from the terminating user?	
	Check: Are the media streams terminated after the 200 OK CANCEL/BYE	
	was sent?	
	Repeat this test in reverse direction.	

Reference 6.11.2/[24] SELECTION EXPRESSION [Network B] SE 17 Test purpose SIP-I support. Called number is not allocated in the PSTN/PLMN network.	Test case number	SS_unsucc_012	
SELECTION EXPRESSION [Network B] SE 17	Test case group	BCALL/unsuccessful	
SIP-I support. Called number is not allocated in the PSTN/PLMN network. Ensure that, when calling to an unallocated number in the PSTN/PLMN part of network B and ISUP - SIP-I interworking applies in Network B, the network initiate call clearing to the calling user with a 404 Not Found message. A ISUP REL message is encapsulated and the Cause value indicator is set to '1'. Configuration The called user number is not assigned to the PSTN/PLMN part in Network B 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)	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 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)	SELECTION EXPRESSION	[Network B] SE 17	
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)	Test purpose	SIP-I support. Called number is not allocated in the PSTN/PLMN network.	
network B and ISUP - SIP-I interworking applies in Network B, the network initiate call clearing to the calling user with a 404 Not Found message. A ISUP REL message is encapsulated and the Cause value indicator is set to '1'. Configuration The called user number is not assigned to the PSTN/PLMN part in Network B 404: Reason: Q.850;cause=1 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B)			
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)			
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)			
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)			
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)			
Reason: Q.850;cause=1 (optional)	Configuration	The called user number is not assigned to 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: 1[any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B)	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 SIP (Network B)			
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)		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 SIP (Network B)			
Content-Disposition: signal;handling=required REL Cause value: 1[any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B)			
REL Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B)			
Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B)		Content-Disposition: signal;nandling=required	
Cause value: 1 [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B)		DEI	
[any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B)			
Message flow SIP (Network A) Interconnection Interface SIP (Network B)		Cause value. 1	
Message flow SIP (Network A) Interconnection Interface SIP (Network B)		lany boundary namel	
SIP (Network A) Interconnection Interface SIP (Network B)	Message flow	[, 40000]	
INVITE -		Interconnection Interface SIP (Network B)	
	,	INVITE -	
← 404 Not Found(REL)		← 404 Not Found(REL)	
ACK →		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	
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 number	SS_unsucc_013	
Test case group	BCALL/unsuccessful	
Reference	6.11.2/[24]	
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47	
Test purpose	SIP-I support. The called user is busy.	
	Ensure that, when the called user in the PSTN/PLMN part of Network B and	
	ISUP - SIP-I interworking applies in Network B is busy, the network initiates call	
	clearing to the calling user with a 486 Busy Here message. A ISUP REL	
	message is encapsulated and the Cause value indicator is set to '17'.	
Configuration	The called user is busy in the PSTN/PLMN part in Network B	
SIP Parameter	486:	
	Reason: Q.850;cause=17 (optional)	
	Content-Type: multipart/mixed;boundary=[any boundary name]	
	[any boundary name]	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	REL	
	Cause value: 17	
	Cadoo valao. 17	
	[any boundary name]	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
	INVITE →	
	← 486 Busy Here(REL)	
	ACK →	
Comments	Establish a communication from network A to Network B, user B in the	
	PSTN/PLMN part of Network B is busy.	
	Check: Is a 486 Busy Here sent from Network B to Network A?	
	Check: Is a ISUP REL encapsulated and the Cause value indicator is set to '17'?	
	Check: If a Reason header is present, is the cause value equal to the value in	
	the Cause value of the encapsulated ISUP REL?	
	Repeat this test in reverse direction.	

Test case number	SS_unsucc_014
Test case group	BCALL/unsuccessful
Reference	6.11.2/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	SIP-I support. The called user rejects the call.
	Ensure that, when the called user in the PSTN/PLMN part of Network B and ISUP - SIP-I interworking applies in Network B rejects the communication setup, the network initiates call clearing to the calling user with a 480 Temporarily Unavailable final response. A ISUP REL message is encapsulated and the Cause value indicator is set to '21'.
Configuration	
SIP Parameter	480: Reason: Q.850;cause=21 (optional) 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
Test purpose	SIP-I support. Call clearing due to no answer from the called user initiated
rest purpose	by the calling user.
	Ensure when the early dialogue is not confirmed by the called user, the calling user located in the PSTN/PLMN part of Network A and ISUP - SIP-I interworking applies in Network A initiates call clearing to the called user with CANCEL or BYE. An ISUP REL message is encapsulated in the BYE request and the Cause
	value indicator is set to '16'.
Configuration	
SIP Parameter	480:
	Reason: Q.850;cause=16 (optional) Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required
	REL
	Cause value: 16
	Cause value. 10
	[any boundary name]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 180 Ringing
CASE A	← 180 Ringing
CASE A	CANCEL →
	€ 200 OK CANCEL
	← 487 Request Terminated
	ACK
	NON 2
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 Ensure w originating calling us interworki or BYE to	pport. Call clearing due to no iginating network. then the early dialogue is not concern the call clear in the call clear in the PSTN/PLM ing applies in Network A and the concern the call clear in the PSTN/PLM ing applies in Network A and the concern the call clear in the PSTN/PLM ing applies in Network A and the concern the co	onfirmed by the called user, the ing after timeout of ISUP timer T9 if the N part of Network A and ISUP - SIP-I ne originating network sends a CANCEL message is encapsulated in the BYE of the 140'
Configuration	request a	ind the Cause value indicator i	3 361 10 19 .
SIP Parameter	480:		
SIF Farameter	Reason: 0	Q.850;cause=19 (optional) ontent-Type: multipart/mixed;boany boundary name] ontent-Type: application/isup;vontent-Disposition: signal;hand	
	_	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: Check: Check:	a communication from network to communication setup. It is a CANCEL or BYE request Is a ISUP REL encapsulated Is the Cause value of the encaption of the cause value of the encaption of the cause value of the encaption of the cause value of the encaption of th	is sent by the originating network? in a BYE request? capsulated REL set to '19'? it, is the cause value equal to the value in sulated ISUP REL? I send from the terminating user? ated after the 200 OK CANCEL/BYE

7.1.5 Test purposes for Supplementary services

7.1.5.1 Test purposes for OIP

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

Test case number	CC oin 002
	SS_oip_002
Test case group	SIP-SIP/Service/OIP
Reference	5.2.6.3/[2]
SELECTION EXPRESSION	
Test purpose	P-Preferred-Identity received, no match with the set of registered public identities. The terminating user receives the default public user identity of the originating user.
	In case the preconditions are fulfilled to provide the terminating UE with originating identification information without preventing the presentation, ensure that an identity information in the P-Preferred-Identity header is provided by the originating UE, does not match with the set of registered public identities of the originating UE the terminating user receives a P-Asserted-Identity based on the default public user identity associated with the originating UE identifies the originator of the session.
Configuration	
SIP Parameter	INVITE P-Asserted-Identity= default public user identity
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →
Comments	Check: Is the P-Asserted-Identity set to the default public user identity? Check: Is optional a second P-Asserted-Identity header present as a 'tel' URI with a public user identity? Check: Is the user parameter is set to phone? Check: Is the P-Preferred-Identity header not present?
	Repeat this test in reverse direction. Repeat this test with all relevantend devices.

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

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

Test case number	SS_oip_005
Test case group	SIP-SIP/Service/OIP
Reference	4.5.2.4/[7]
SELECTION EXPRESSION	SE 18 AND SE 19
Test purpose	Special arrangement exists.
	The special arrangement exists (no screening of user provided information). The network does not attempt to match the information in the From header with the set of registered public identities of the originating user. The From header field is transparently transported to the terminating user.
Configuration	Special arrangement for the originating user exists
SIP Parameter	INVITE
	From= original value
	P-Asserted-Header=[any registered public user identity]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →
Comments	Check: Is the From header URI set to original value sent by the user? Check: Is the P-Asserted-Identity set to any registered public user identity? Check: Is the user parameter is set to phone? Repeat this test in reverse direction. Repeat this test with all relevantend devices.

Test case number	SS_oip_006
Test case group	SIP-SIP/Service/OIP
Reference	7.1.3/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 52
Test purpose	SIP-I support. ISUP Calling party number presentation allowed in the
	encapsulated IAM.
	Ensure when BICC/ISUP - SIP-I interworking applies in the originating network
	the BICC/ISUP IAM is encapsulated in the INVITE request. The P-Asserted-
	Identity header field is derived from the Calling party number in the encapsulated
	IAM. The 'Presentation restriction' indicator in the encapsulated IAM is set to
	'allowed' no Privacy value 'id' is present in the INVITE request.
Configuration	
SIP Parameter	INVITE
	P-Asserted-Identity=[derived from the ISUP calling party number]
	Content-Type: multipart/mixed;boundary=[any boundary name]
	[any hayndary nama]
	[any boundary name]
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required
	Content-Disposition. Signal, nanding=required
	IAM
	Calling party number
	Screening indicator
	Network provided or user provided, verified and
	passed
	Presentation restriction
	allowed
	Address signal
	[any boundary name]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE(IAM) →
Comments	Check: Is a BICC/ISUP IAM encapsulated in the in the INVITE request?
	Check: Is the Calling party number present in the encapsulated IAM and the
	screening indicator is set to 'Network provided' or 'user provided,
	verified and passed' and the Presentation restriction indicator is set to
	'allowed'? Check: Is the P-Asserted-Identity header field derived from the Calling party
	31,
	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.
	Incheat this test in reverse direction.

Test case number	SS_oip_007
Test case group	SIP-SIP/Service/OIP
Reference	7.1.3/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 52
Test purpose	SIP-I support. ISUP Additional Calling party number presentation allowed
rest purpose	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	[any bodinary name]
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(IAM) →
Comments	Check: Is a BICC/ISUP IAM encapsulated in the in the INVITE request?
	Check: Is the Calling party number present in the encapsulated IAM and the screening indicator is set to 'Network Provided' and the Presentation
	restriction indicator is set to 'allowed'? Check: Is the P-Asserted-Identity header field derived from the Calling party
	number in the encapsulated IAM?
	Check: Is a Generic number parameter, Number Qualifier Indicator set to
	Additional calling party number present and the screening indicator
	is set to 'user provided, not verified' and the Presentation restriction
	indicator is set to 'allowed'?
	Check: Is the From header field derived from the Additional calling party number in the encapsulated IAM?
	Check: Is the value 'id' not present in the Privacy header field (if included)?
	Repeat this test in reverse direction.
	Trapagrama (agram tarataa amaanam

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 →
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	ICC oir 0	002		
	SS_oir_002			
Test case group	SIP-SIP/Service/OIR			
Reference	4.3.2, 4.5.2.4/[7]			
SELECTION EXPRESSION	SE 20 AND SE 25			
Test purpose	Communication forwarding unconditional, served user subscribes OIR.			
	user B is	A and user C are in network B and in network A and is provided with 0 ted-to user"=Yes.	d user C is provided with OIP. The CFU "diverting number is released to	
	permane unconditi diverted-		calls user B, the call is forwarded	
	header n	or in the To header.		
Configuration		user subscribes to the OIR service	9	
SIP Parameter	INVITE : r	no history entry present		
	INVITE :			
	History-Info header:			
	<sip:userb@networka?privacy=history>;index=1,</sip:userb@networka?privacy=history>			
	<sip: userc@networkb;cause="302">;index=1.1</sip:>			
Message flow				
SIP (Network A)		Interconnection Interface	SIP (Network B)	
	+	INVITE		
	С	FU is performed in Network A		
		INVITE →		
		Apply post test routine		
Comments	Check:	No History-Info header is received		
	Check:	Is the Privacy value history is esc	aped in the hi-targed-to-uri of the	
		diverting user in Network A?		
	Repeat this test in reverse direction.			
	Repeat th	his test with all chosen end devices	S.	

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		
0	INVITE request.		
Configuration	INDUCTE.		
SIP Parameter	INVITE		
	P-Asserted-Identity=[derived from the ISUP calling party number]		
	Privacy: id Content-Type: multipart/mixed;boundary=[any boundary name]		
	Content-Type: multipar/mixed,boundary=[arry boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Type: application/isup,version=itu-ts2 Content-Disposition: signal;handling=required		
	Content-Disposition, signal, nationing-required		
	IAM		
	Calling party number		
	Screening indicator		
	Network provided or user provided, verified and		
	passed		
	Presentation restriction		
	restricted		
	Address signal		
	[any have down name]		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
Cir (ricinicini,	INVITE(IAM)		
Comments	Check: Is a BICC/ISUP IAM encapsulated in the in the INVITE request?		
	Check: Is the Calling party number present in the encapsulated IAM and the		
	screening indicator is set to 'Network provided' or 'user provided,		
	verified and passed' and the Presentation restriction indicator is set to		
	'restricted'?		
	Check: Is the P-Asserted-Identity header field derived from the Calling party		
	number in the encapsulated IAM?		
	Check: Is the value 'id' present in the Privacy header field?		
	Repeat this test in reverse direction.		

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

7.1.5.3 Test purposes for TIP

Test case number	SS_tip_0	001		
Test case group	SIP-SIP/	Service/TIP		
Reference	5.2.6.4/[8	8]		
SELECTION EXPRESSION				
Test purpose	Originating user receives the identity of the terminating user.			inating user.
	Ensure in case the preconditions are fulfilled to provide the originating UE with terminating identification information without preventing the presentation, the originating UE receives in a 1xx or 200 SIP response a P-Asserted-Identity header field with a valid public user identity of the terminating UE.			
Configuration				
SIP Parameter	18x/200	OK INVITE		
		P-Asserted-Identity:		
Message flow SIP (Network A)		Interconnection Interface INVITE	→	SIP (Network B)
CASE A	←	180 Ringing		
CASE B	←	183 Session Progress		
CASE C	← 20	00 OK INVITE(P-Asserted-Identity) Apply post test routine		
Comments		Is the P-Asserted-Identity is present Progress or in a 200 OK INVITE? his test in reverse direction. 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		
	was received.		
Configuration	Special arrangement for the terminating user exists		
SIP Parameter	INVITE		
	Supported: from-change		
	200 OK INVITE		
	P-Asserted-Identity:		
	UPDATE		
	From: (second user identity)		
Message flow	(**************************************		
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE ->		
	← 180 Ringing		
	← 200 OK INVITE(P-Asserted-Identity)		
	ACK → UPDATE (From)		
	← UPDATE (From) 200 OK UPDATE →		
	Apply post test routine		
Comments	Check: Is the 'from-change' tag present in the Supported header of the initial		
	INVITE request?		
	Check: Is the P-Asserted-Identity is present in a 180 Ringing or 183 Session		
	Progress or in a 200 OK INVITE?		
	Check: Is the 'from-change' tag present in the supported header of the		
	provisional (18x) or final (200 OK) response?		
	Check: Is an UPDATE request sent by the terminating user containing a From		
	header field set to the value send by the terminating user? Repeat this test in reverse direction.		
	Repeat this test with all chosen end devices.		
<u> </u>	repeat the test that an officer office devices		

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

Test case number	SS_tip_004		
Test case group	SIP-SIP/Service/TIP		
Reference	6.7/[24]		
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 53		
Test purpose	SIP-I support. The Connected number presentation allowed is present in		
	the encapsulated 200 OK.		
	Ensure that on receipt of a 200 OK INVITE to establish a confirmed dialogue an		
	ANM is encapsulated if SIP-I - BICC/ISUP interworking is applicable in Network		
	B. The Address presentation restriction indicator is set to 'allowed'. The		
	screening indicator is set to Network provided or user provided, verified and		
	passed.		
Configuration			
SIP Parameter	200 OK INVITE		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	ANM		
	Connected number		
	Screening indicator Network provided or user provided, verified and		
	passed		
	Address presentation restriction		
	allowed		
	Address signal		
	1.000.000 0.9.00		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(IAM) →		
	← 180 Ringing(ACM)		
	← 200 OK INVITE(ANM)		
	Ack →		
Comments	Apply post test routine Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final		
Comments	·		
	response? Check: Is the Screening indicator in the encapsulated ANM set to 'Network		
	provided' or 'user provided, verified and passed'?		
	Check: Is the Address presentation restriction indicator in the encapsulated		
	ANM set to allowed?		
	Repeat this test in reverse direction.		
	proposit tino toot in reverse unconon.		

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

7.1.5.4 Test purposes for TIR

Tool sees would be	00 45- 0	04		
Test case number	SS_tir_0			
Test case group	SIP-SIP/	Service/TIR		
Reference	4.5.2.9/[8	8]		
SELECTION EXPRESSION	SE 23			
Test purpose	Originating user does not receive the identity of the terminating user.			
		hat, when the preconditions are ful-		revent the presentation of
		nating user identity at the originatir		
		nating UE receives, in any non-100		
	,	neader field is set to "id" and no P-/	Asserted-	Identity header field is
	present.			
Configuration	The term	ninating 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		
0405 B	-	100 O : D		
CASE B	←	183 Session Progress		
CASE C	← 20	OO OK INIVITEID Asserted Identity		
CASE C	~ 20	00 OK INVITE(P-Asserted-Identity) Apply post test routine	1	
Comments	Check:	Is the P-Asserted-Identity is pres	ont in the	provisional (19y) or final
Comments	Cileck.	(200 OK) response?		provisional (10x) of ilital
	Check:	Is the Privacy header in the provi	sional (19	(200 OK)
	Office.	response is set to 'id'?	Sional (10) or illiar (200 OK)
	Reneat ti	his test in reverse direction.		
		his test with all chosen end devices		
	וו זבטבמו וו	ina teat with all thosen end device:	.	

Test case number	SS_tir_002		
Test case group	SIP-SIP/Service/TIR		
Reference	6.7/[24]		
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 53		
Test purpose	SIP-I support. The Connected number presentation allowed is present in		
	the encapsulated 200 OK.		
	Ensure that on receipt of a 200 OK INVITE to establish a confirmed dialogue an		
	ANM is encapsulated if SIP-I - BICC/ISUP interworking is applicable in Network		
	B. The Address presentation restriction indicator is set to 'restricted'. The		
	screening indicator is set to 'Network provided' or 'user provided, verified and		
	passed'.		
Configuration			
SIP Parameter	200 OK INVITE		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any houndary name]		
	[any boundary name] Content-Type: application/isup;version=itu-t92		
	Content-Type: application/isup,version=itu-is2 Content-Disposition: signal;handling=required		
	Content-Disposition: Signal, nanuling=required		
	ANM		
	Connected number		
	Screening indicator		
	Network provided or user provided, verified and		
	passed		
	Address presentation restriction		
	restricted		
	Address signal		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(IAM) →		
	← 180 Ringing(ACM)		
	← 200 OK INVITE(ANM)		
	ACK →		
_	Apply post test routine		
Comments	Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final		
	response?		
	Check: Is the Screening indicator in the encapsulated ANM set to 'Network		
	provided' or 'user provided, verified and passed'?		
	Check: Is the Address presentation restriction indicator in the encapsulated ANM set to allowed?		
	Repeat this test in reverse direction.		

Test case number	SS tir 00	03		
Test case group		Service/TIR		
Reference	6.7/[24]			
SELECTION EXPRESSION		B] SE 17 AND SE 47 AND SE 53		
Test purpose	SIP-I sup	oport. The additional connected number restricted is present in the lated 200 OK.		
	ANM is e B. A Gen 'additiona not verifie A Conne	nat on receipt of a 200 OK INVITE to establish a confirmed dialogue an incapsulated if SIP-I - BICC/ISUP interworking is applicable in Network eric number parameter is present the Number qualifier indicator set to all connected number the Screening indicator is set to 'user provided, ed' and the Address Presentation Restricted is set to 'restricted'. coted number parameter is present the Screening indicator is set to provided' and the Address Presentation Restricted indicator is set to d'.		
Configuration		inating user in the PSTN/PLMN part of Network B is subscribed to the o screening option'		
SIP Parameter	200 OK I P-			
	C	any boundary name] ontent-Type: application/isup;version=itu-t92 ontent-Disposition: signal;handling=required		
		ANM		
		Connected number		
	Screening indicator Network provided or user provided, verified and passed Presentation restriction restricted			
		Address signal		
	Generic number Number Qualifier Indicator Additional calling party number			
		Screening indicator		
		user provided, not verified		
		Address Presentation Restricted		
		restricted		
		Address signal		
	[any boundary name]		
Message flow		Interconnection Interfere		
SIP (Network A)		Interconnection Interface SIP (Network B)		
	_	INVITE(IAM) →		
	-	180 Ringing(ACM) 200 OK INVITE(ANM)		
	•	ACK →		
		Apply post test routine		
Comments	Check:	Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final		
		response?		
	Check:	Is a Generic number parameter present in the encapsulated ANM?		
	Check:	Is the Number Qualifier Indicator of the Generic number set to		
		'additional connected number'?		
	Check:	Is the Screening indicator of the Generic number set to 'user provided,		
	<u> </u>	not verified'?		
	Check:	Is the Address presentation restriction indicator in the Generic number		
	Da =	set to 'allowed'?		
	Repeat th	nis test in reverse direction.		

7.1.5.5 Communication Hold (HOLD)

Test case number	SS_hold_001		
Test case group	SIP-SIP/Service	ce/HOLD	
Reference	4.5.2.1/[17]		
SELECTION EXPRESSION	SE 24		
Test purpose	Hold the session the media stream was previously set to sendrecv.		
	UPDATE requent attribute "a=se		s done containing the SDP with the esting the hold session <i>receives</i> 200
Configuration			
SIP Parameter			
Message flow SIP (Network A)		erconnection Interface ned session already exists	SIP (Network B)
CASE A		INVITE(<mark>sendonly</mark>) 0 OK INVITE (recvonly) ACK	→ →
CASE B		UPDATE(<mark>sendonly</mark>) OK UPDATE (recvonly) ply post test routine	→
Comments	Check: Is the	e user in network A able to se	et the session on hold by sending an he version parameter in the SDP 'o'

Test case number	SS_hold_	_002		
Test case group	SIP-SIP/S	Service/HOLD		
Reference	4.5.2.1/[1	7]		
SELECTION EXPRESSION	SE 24	•		
Test purpose	Hold the	session the media stream was p	previously set to recvonly.	
	Ensure th	nat the UE A requesting <mark>hold</mark> of the	session stops sending media and	
	sends an INVITE or UPDATE request to hold the session. Hold is done			
			ctive". The UE A after requesting to	
		ne held session receives 200 OK fi	nal response containing the SDP	
	with the a	attribute "a= <mark>inactive.</mark> "		
Configuration				
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface	SIP (Network B)	
		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 (<mark>sendonly</mark>)		
CASE B	•	200 OK INVITE (recvonly)	→	
	←	ACK	7	
	•	UPDATE(<mark>inactive</mark>)	→	
	←	200 OK UPDATE (inactive)		
	•	200 OR OF BITTE (mactive)		
CASE C	(UPDATE (<mark>sendonly</mark>)		
		200 OK UPDATE (recvonly)	→	
		INVITE (inactive)	→	
	←	200 OK INVITE (inactive)		
		ACK `	→	
CASE D	←	UPDATE (<mark>sendonly</mark>)		
		200 OK UPDATE (recvonly)	→	
		UPDATE(<mark>inactive</mark>)	→	
	←	200 OK UPDATE (inactive)		
	1	Apply post test routine		
Comments	Check:		et the session on hold by sending an	
		•	the version parameter in the SDP 'o'	
	01 1	line is incremented?	-4.46	
	Check:		et the session on hold by sending an	
		•	the version parameter in the SDP 'o'	
	Donast "	line is incremented?		
	repeat tr	nis test in reverse direction.		

Test case number	SS_hold_003			
Test case group	SIP-SIP/Service/HOLD			
Reference	4.5.2.1/[17]			
SELECTION EXPRESSION	SE 24			
Test purpose	Resume the session the media stream was previously set to sendonly.			
	Engine that the LIE A is requested to request			
		Ensure that the UE A is requested to resume the session with user B the UE-A starts sending media and sends an INVITE or UPDATE request to resume the session with the attribute "a=sendrecv in the SDP. The UE A after requesting to resume the held session receives 200 OK final response and optionally the attribute "a=sendrecv in the SDP. The a=sendrecv attribute is the default value		
	therefore	the attribute can be omitted.		
Configuration				
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface	SIP (Network B)	
	A c	onfirmed session already exists		
CASE A	-	INVITE (sendonly)	→	
	←	200 OK INVITE (recvonly) ACK	→	
		INVITE (<mark>sendrecv</mark>)	→	
	←	200 OK INVITE (sendrecv)	*	
	-	ACK	→	
CASE B		INVITE (<mark>sendonly</mark>)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	•	UPDATE (sendrecv)	→	
	←	200 OK UPDATE (sendrecv)		
CASE C		UPDATE (<mark>sendonly</mark>)	→	
CAGE C	←	200 OK UPDATE (recvonly)	•	
	_	INVITE (sendrecv)	→	
	←	200 OK INVITE (sendrecv)		
		ACK	→	
			_	
CASE D	-	UPDATE (sendonly)	→	
	←	200 OK UPDATE (recvonly) UPDATE (<mark>sendrecv</mark>)	→	
	←	200 OK UPDATE (sendrecv)	7	
	•	Apply post test routine		
Comments	Check:		t the session on hold by sending an	
			ne version parameter in the SDP 'o'	
		line is incremented?	-	
	Check:	Is the user in network A able to ret		
			ne version parameter in the SDP 'o'	
		line is incremented? The absence	of the 'sendrecv' attribute is the	
	Donact ti	default value.		
	Repeat	nis test in reverse direction.		

Test case number	SS_hold_	004	
Test case group	SIP-SIP/Service/HOLD		
Reference	4.5.2.1/[1		
SELECTION EXPRESSION	SE 24	<i>'</i>]	
	Resume the session the media stream was previously set to inactive.		
Test purpose	Resume	the session the media stream was	s previously set to mactive.
	The See	sion is in the "inactive" state. Ensure	that the LIE A is requesting to
		ne session with user B the UE-A ser	
		ne session with user B the OE-A ser ne session with the attribute <mark>"a=recv</mark>	
		g to resume the held session <i>receiv</i>	
		y the attribute <mark>"a=sendonly</mark> in the SD	
Configuration	optionally	r the attribute a=sendonly in the SD	۲.
Configuration SIP Parameter			
Message flow		Interconnection Interface	CID (Notwork D)
SIP (Network A)	A	Interconnection Interface	SIP (Network B)
CASE A		onfirmed session already exists	
CASE A	←	INVITE(sendonly)	
	-	200 OK INVITE (recvonly)	→
	←	ACK	
	-	INVITE(inactive)	→
	←	200 OK INVITE (inactive)	
		ACK	→
	-	INVITE (recvonly)	→
	←	200 OK INVITE (sendonly)	
		ACK	→
CASE B	_	INIVITE (a a sa da salva)	
CASE B	←	INVITE(sendonly)	_
	_	200 OK INVITE (recvonly)	→
	←	ACK	_
	_	UPDATE(inactive)	→
	←	200 OK UPDATE (inactive)	_
	←	INVITE (recvonly)	→
	~	200 OK INVITE (<mark>sendonly</mark>)	→
		ACK	7
CASE C	←	UPDATE (sendonly)	
CAGE C	•	200 OK UPDATE (recvonly)	_
		INVITE(inactive)	→
	←	200 OK INVITE (inactive)	
	•	ACK	→
		UPDATE (<mark>recvonly</mark>)	<u> </u>
	←	200 OK UPDATE (sendonly)	
	•	200 OR OF DATE (Schooling)	
CASE D	←	UPDATE (sendonly)	
0.102.5	•	200 OK UPDATE (recvonly)	→
		UPDATE(inactive)	→
	←	200 OK UPDATE (inactive)	
	_	UPDATE (recvonly)	→
	←	200 OK UPDATE (sendonly)	-
	•	Apply post test routine	
Comments	Check:	Is the user in network B able to set	the session on hold by sending an
		INVITE or UPDATE request and th	
		line is incremented?	- 1.1.1.0 parameter in the CD .
	Check:	Is the user in network A able to set	the session on hold by sending an
	G.I.GGIKI	INVITE or UPDATE request and th	
		line is incremented?	c research parameter in the ODI O
	Check:	Is the user in network A able to reti	rieve the session by sending an
	JJUIL.	INVITE or UPDATE request and th	
		line is incremented?	c research parameter in the ODI O
	Repeat th	nis test in reverse direction.	
L	i ropeat ii	no toot iii iovoioo diigotioii.	

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 pr	reviously set to sendrecv.	
	Ensure that the UE A receives an INVITE or UPDATE request to hold the session and stops sending media. Hold is done containing the SDP with the attribute "a=sendonly". The UE A after resuming the held session sends a 200 OK final response containing the SDP with the attribute "a=recvonly".		
Configuration			
SIP Parameter			
Message flow SIP (Network A)	Interconnection Interface A confirmed session already exists	SIP (Network B)	
CASE A	 ← INVITE(sendonly) 200 OK INVITE(recvonly) ← ACK 	→	
CASE B	← UPDATE(sendonly) 200 OK UPDATE (recvonly) Apply post test routine	→	
Comments		the session on hold by sending an e version parameter in the SDP 'o'	

Test case number	SS_hold	006	
Test case group	SIP-SIP/Service/HOLD		
Reference	4.5.2.1/[1	l7]	
SELECTION EXPRESSION	SE 24	-	
Test purpose	Hold the session the media stream was previously set to sendonly.		
	The Session is in the "sendonly" state. Ensure that the UE A receives an INVITE		
	or UPDATE request to hold the session and stops sending media. Hold is done		
	containing the SDP with the attribute "a=inactive". The UE A after receiving the hold session sends 200 OK final response containing the SDP with the attribute		
	"a=inacti		containing the SDP with the attribute
Configuration	a= <mark>macu</mark>	ve .	
SIP Parameter			
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
on (Notwork A)	A c	onfirmed session already exists	on (notwork b)
CASE A	, , ,	INVITE(<mark>sendonly</mark>)	→
	(200 OK INVITE (recvonly)	
		ACK `	→
	←	INVITE (inactive)	
		200 OK INVITE (<mark>inactive</mark>)	→
	←	ACK	
CASE B		INVITE(sendonly)	→
CASE B	←	200 OK INVITE (recvonly)	7
		ACK	→
	←	UPDATE (inactive)	
		200 OK UPDATE (inactive)	→
		,	
CASE C	_	UPDATE (<mark>sendonly</mark>)	→
	(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 (inactive)	→
		Apply post test routine	
Comments	Check:		et the session on hold by sending an
	01-1-1	INVITE or UPDATE request?	et the energies on health
	Check:		et 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.		
	Inchear	ino test in reverse unection.	

Test case number	SS_hold_	_007		
Test case group	SIP-SIP/S	Service/HOLD		
Reference	4.5.2.1/[1	7]		
SELECTION EXPRESSION	SE 24			
Test purpose	Resume	the session the media stream w	as previously set to recvonly.	
	F	E de la LIEA de LINNUTE LINNUTE		
		Ensure that the UE A receives an INVITE or UPDATE request requesting to		
		resume the session with user B, the UE-A starts sending media. Resume is done containing the SDP with the attribute "a=sendrecv". The UE A after receiving the		
			response containing the SDP with	
		ute "a= <mark>sendrecv</mark> ". The a=sendrecv		
		the attribute can be omitted.	attribute is the default value	
Configuration	uicicioic	the attribute can be offitted.		
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface	SIP (Network B)	
on (Notwork 71)	A co	onfirmed session already exists		
CASE A	←	INVITE (sendonly)		
		200 OK INVITE(recvonly)	→	
	(ACK		
	←	INVITE(sendrecv)		
		200 OK INVÎTE(sendrecv)	→	
	←	ACK		
CASE B	←	UPDATE (sendonly)		
		200 OK UPDATE (<mark>recvonly</mark>)	→	
	←	UPDATE (sendrecv)		
		200 OK UPDATE (<mark>sendrecv</mark>)	→	
		Apply post test routine		
Comments	Check:		et the session on hold by sending an	
			the version parameter in the SDP 'o'	
	<u>.</u>	line is incremented?		
	Check:		etrieve the session by sending an	
			the version parameter in the SDP 'o'	
	line is incremented?			
	repeat tr	nis test in reverse direction.		

Test case number	SS_hold	008	
Test case group		Service/HOLD	
Reference	4.5.2.1/[1		
		7]	
SELECTION EXPRESSION	SE 24 Resume the session the media stream was previously set to inactive.		
Test purpose	Resume	the session the media stream wa	is previously set to inactive.
	T 1 0		
		sion is in the "inactive" state. Ensure	
		TE request requesting to resume th	
		nding media. Resume is done conta	
			tesume of the session sends 200 OK
		onse containing the SDP with the a	
	a=sendre	ecv attribute is the default value ther	refore the attribute can be omitted.
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
,	A c	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	
	•	/ tort	
CASE B		INVITE (sendonly)	→
CAGE B	←	200 OK INVITE (recvonly)	
	•	ACK	→
	←	UPDATE (<mark>inactive</mark>)	7
	~		→
	←	200 OK UPDATE (inactive)	7
	~	UPDATE (recvonly)	_
		200 OK UPDATE (<mark>sendonly</mark>)	→
CASE C		LIDDATE (condents)	_
CASE C	_	UPDATE (sendonly)	→
	-	200 OK UPDATE (recvonly)	
	~	INVITE (inactive)	
	_	200 OK INVITE (inactive)	→
	(ACK	
	~	INVITE (recvonly)	
	,	200 OK INVITE (sendonly)	→
	(ACK	
CASE D		LIDDATE (
CASE D	,	UPDATE (sendonly)	→
	(200 OK UPDATE (recvonly)	
	←	UPDATE (inactive)	
	_	200 OK UPDATE (inactive)	7
	←	UPDATE (<mark>recvonly</mark>)	_
		200 OK UPDATE (sendonly)	→
		Apply post test routine	
Comments	Check:		t the session on hold by sending an
			ne version parameter in the SDP 'o'
		line is incremented?	
	Check:		t the session on hold by sending an
		INVITE or UPDATE request and the	ne version parameter in the SDP 'o'
		line is incremented?	
	Check:	Is the user in network B able to ret	trieve the session by sending an
		INVITE or UPDATE request and the	ne version parameter in the SDP 'o'
		line is incremented?	
	Repeat tl	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			
Test purpose		dia etroam was proviously sot to		
rest purpose	Resume the session on both sides the media stream was previously set to inactive.			
	inactive.			
	The Consider in the Historian state. Frances	de et the elle A is assured as 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 se-			
	"a= <mark>sendonly</mark> in the SDP. The UE A after requ			
	receives 200 OK final response containing the			
	"a= <mark>recvonly</mark> . The UE B after requests to resu			
	final response containing the SDP with the att	ribute "a= <mark>sendrecv"</mark> . The		
	a=sendrecv attribute is the default value there	fore the attribute can be omitted.		
Message flow				
SIP (Network A)	Interconnection Interface	SIP (Network B)		
,	A confirmed session already exists	` ,		
CASE A	INVITE(sendonly)	→		
	← 200 OK INVITE (recvonly)	-		
	ACK	→		
	← INVITE(inactive)	-		
	200 OK INVITE (inactive)	→		
	← ACK			
	INVITE(sendonly)	→		
	← 200 OK INVITE (recvonly)	7		
	ACK	→		
	← INVITE(sendrecv)	7		
	200 OK INVITE (sendrecv)	→		
	← ACK			
0405 5	INDUTE:			
CASE B	INVITE(sendonly)	→		
	← 200 OK INVITE (recvonly)	_		
	ACK	→		
	← UPDATE (inactive)	_		
	200 OK UPDATE (inactive)	→		
	INVITE(<mark>sendonly</mark>)	→		
	← 200 OK INVITE (recvonly)			
	ACK	→		
	← UPDATE (sendrecv)			
	200 OK UPDATE (sendrecv)	→		
CASE C	UPDATE (sendonly)	→		
	← 200 OK UPDATE (recvonly)			
	← INVITE(inactive)			
	200 OK INVITE (inactive)	→		
	← ACK			
	UPDATE (sendonly)	→		
	← 200 OK UPDATE (recvonly)	-		
	ACK	→		
	← INVITE(sendrecv)			
	200 OK INVITE (sendrecv)	→		
	← ACK			
	ACK			
CASE D	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			
	• • • •			

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

Test case number	SS_hold_010			
Test case group	SIP-SIP/Service/HOLD			
Reference	4.5.2.1/[17]			
SELECTION EXPRESSION	SE 24			
Test purpose	Resume the session on both sides the m	edia stream was previously set to		
	inactive.			
	The Session is in the "inactive" state. Ensure that the UE A receives an INVITE			
	or UPDATE to resume the session with user B, the UE-A starts sending media.			
	Resume is done containing the SDP with the			
	after receiving the Resume of the session se			
	containing the SDP with the attribute "a=sendonly". The UE A after requests to			
	resume the session receives 200 OK final re			
	attribute "a=sendrecv. The UE B after receiving the Resume of the session sends 200 OK final response containing the SDP with the attribute "a=sendrecv".			
	The a=sendrecv attribute is the default value	e therefore the attribute can be		
	omitted.			
Configuration				
SIP Parameter				
Message flow	India a constant de la constant de	CID (ALL)		
SIP (Network A)	Interconnection Interface	SIP (Network B)		
0405.4	A confirmed session already exists			
CASE A	← INVITE(sendonly)			
	200 OK INVITE (recvonly)	→		
	← ACK			
	INVITE(inactive) ← 200 OK INVITE (inactive)	→		
	← 200 OK INVITE (inactive) ACK	→		
	← INVITE(sendonly)	7		
	200 OK INVITE (recvonly)	→		
	← ACK	7		
	INVITE(sendrecv)	→		
	← 200 OK INVITE (sendrecv)			
	ACK	→		
	Hore			
CASE B	← INVITE(sendonly)			
	200 OK INVITE (recvonly)	→		
	← ACK `			
	UPDATE (inactive)	→		
	200 OK UPDATE (inactive)			
	← INVITE(sendonly)			
	200 OK INVITE (<mark>recvonly</mark>)	→		
	← ACK			
	UPDATE (<mark>sendrecv</mark>)	→		
	← 200 OK UPDATE (sendrecv)			
CASE C	← UPDATE (sendonly)			
	200 OK UPDATE (recvonly)	→ →		
	INVITE(inactive) ← 200 OK INVITE (inactive)	7		
	,	→		
	ACK ← UPDATE (sendonly)	7		
	200 OK UPDATE (recvonly)	→		
	INVITE(sendrecv)	→		
	← 200 OK INVITE (sendrecv)			
	ACK	→		
	,			
CASE D	← UPDATE (sendonly)			
	200 OK UPDATE (recvonly)	→		
	UPDATE (inactive)	→		
	← 200 OK UPDATE (inactive)			
	← UPDATE (sendonly)			
	200 OK UPDATE (recvonly)	→		
	UPDATE (<mark>sendrecv</mark>)	→		
	← 200 OK UPDATE (sendrecv)			
	Apply post test routine			
				

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

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

Test case number	SS_hold_012			
Test case group	SIP-SIP/Service/HOLD			
Reference	B.10/[24]			
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 54			
Test purpose	SIP-I support. Hold requested by the called user.			
	Ensure that when an INVITE request updates a confirmed session a CPG is			
	encapsulated if SIP-I - ISUP interworking is applicable in Network B. The			
	Generic Notification Indicator parameter is present set to 'hold'. The 'a' attribute			
	is set to 'sendonly' present in the SDP.			
	In the 200 OK INVITE the 'a' attribute is set to 'recvonly' present in the SDP.			
Configuration				
SIP Parameter	INVITE:			
	Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name]			
	a-condonly			
	a=sendonly			
	[any boundary name]			
	[any boundary name] Content-Type: application/isup;version=itu-t92			
	Content-Type: application/isup,version=itu-ts2 Content-Disposition: signal;handling=required			
	CPG			
	Generic notification			
	remote hold			
	[any boundary name]			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	A confirmed session already exists			
CASE A	← INVITE(sendonly, CPG hold)			
	200 OK INVITE (recvonly) → ACK			
Comments	Apply post test routine Establish a session from Network A to Network B			
Comments	The user in the PSTN/PLMN part of Network B places the session on hold.			
	Check: Is a CPG encapsulated in the INVITE request?			
	Check: Is a Generic notification parameter present the Notification indicator			
	set to 'remote hold'?			
	Check: Is the 'a' attribute in the SDP set to 'sendonly'?			
	Check: Is the Version parameter in the SDP incremented?			
	Repeat this test in reverse direction.			

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

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		
0	INVITE request from the Network A.		
Configuration	100 (177)		
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic notification remote hold		
	or remote retrieval		
	[any boundary name]		
Message flow	[any boundary name]		
SIP (Network A)	Interconnection Interface A confirmed session already exists ← INVITE(sendonly) 200 OK INVITE (recvonly) ← ACK		
	INVITE(inactive, CPG hold) → 200 OK INVITE (inactive)		
	ACK →		
	INVITE(<mark>recvonly</mark> , CPG <mark>retrieval</mark>) → £ 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 Chapter to a CDC encorporated 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 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			
0 (INVITE request from the Network A.			
Configuration SIP Parameter	IN OTE			
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name]			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	CPG			
	Generic notification			
	remote hold			
	or remote retrieval			
	[any boundary name]			
Message flow	[any boundary name]			
SIP (Network A)	Interconnection Interface SIP (Network B) A confirmed session already exists			
	← INVITE(sendonly)			
	200 OK INVITE (recvonly) →			
	← ACK			
	INVITE(inactive, CPG hold) →			
	€ 200 OK INVITE (inactive)			
	ACK →			
	← INVITE(sendonly)			
	200 OK INVITE (recvonly) →			
	← ACK			
	INVITE(sendrecv, CPG retrieval) →			
	€ 200 OK INVITE (sendrecv)			
	ACK →			
	Apply post test routine			
Comments	Establish a session from Network A to Network B			
	The user in Network B places the session on hold.			
	Check: Is the 'a' attribute in the SDP set to 'sendonly'?			
	Check: Is the Version parameter in the SDP incremented?			
	The user in Network A places the session on hold			
	Check: Is a CPG encapsulated in the INVITE request?			
	Check: Is a Generic notification parameter present the Notification indicator			
	set to 'remote hold'? Check: Is the 'a' attribute in the SDP set to 'inactive'?			
	Check: Is the Version parameter in the SDP incremented?			
	The user in Network B retrieves the session			
	Check: Is the 'a' attribute in the SDP set to 'sendonly'?			
	Check: Is the Version parameter in the SDP incremented?			
	The user in Network A retrieves the session			
	Check: Is a CPG encapsulated in the INVITE request?			
	Check: Is a Generic notification parameter present the Notification indicator			
	set to 'remote retrieval'?			
	Check: Is the 'a' attribute in the SDP set to 'sendrecy'? Check: Is the Varian parameter in the SDP incremented?			
	Check: Is the Version parameter in the SDP incremented?			
	Repeat this test in reverse direction.			

7.1.5.6 Communication Diversion (CDIV)

7.1.5.6.1 Communication Forwarding Unconditional (CFU)

Test case number	SS_cfu_001			
Test case group	SIP-SIP/Service/CFU			
Reference	4.5.2.6/[9]			
SELECTION EXPRESSION	SE 25			
Test purpose	Commun	nication forwarding uncondition	al, basic r	ules.
	The user A and user C are in Network A. The user B is in network B and is			
	provided	with CFU.		
	Ensure th	nat when user A calls user B, the o	all is forwa	rded 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		SIP (Network B)
		INVITE(Call-ID A-B)	→	
	_	CFU is performed		
	←	INVITE(Call-ID B-C)	_	
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
	,	200 OK INVITE(Call-ID C-B)	→	
	(ACK(Call-ID B-C)		
	~	200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B)	→	
		Communication	7	
		Apply post test routine		
Comments	Check:	CDIV unconditional is successful		
	Check:	In the active call state, ensure th	· -	of speech.
	Check:	Is the P-Asserted-Identity preser		•
	user?			, i i i i i i i i i i i i i i i i i i i
	Repeat this test in reverse direction.			

Test case number	SS_cfu_0	002		
Test case group	SIP-SIP/	SIP-SIP/Service/CFU		
Reference	4.5.2.6/[9	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 25 A	SE 25 AND SE 30		
Test purpose	Commu	Communication forwarding unconditional, no notification.		
Configuration	provided his comn Ensure th C, the or	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.		
Comiguration		Subscription options: Originating user receives notification that his communication has been diverted =		
SIP Parameter				
Message flow SIP (Network A)		Interconnection Interface INVITE(Call-ID A-B) CFU is performed	→	SIP (Network B)
	←	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 interconnection interface. his test in reverse direction.	rwarding in	network B is received at the

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

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

Test case number	SS_cfu_005		
Test case group	SIP-SIP/Service/CFU		
Reference	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 25 AND SE 30		
Test purpose	Communication forwarding unconditional, diverted-to user does not receive the URI of the served user.		
	The user A and user C are in network A. The user B is in network B and is provided with CFU "Served user allows the presentation of his/her URI to the diverted-to user"= No. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user C is not informed of the forwarding number.		
Configuration	Subscription options: Served user allows the presentation of his/her URI to the diverted-to user = No		
SIP Parameter	INVITE: Request line contains ';cause=302' History-Info header: <sip:userb@networkb?privacy=history>;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. 		

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

Test case number	SS_cfu_007		
Test case group	SIP-SIP/Service/CFU		
Reference	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 25 AND SE 30		
Test purpose	Communication forwarding unconditional, full notification.		
	The user A and user C are in network A. The user B is in network B and is provided with CFU Originating user receives notification that his communication has been diverted = Yes and ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, and "Served user allows the presentation of his/her URI to diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number.		
Configuration	Subscription options:		
	 Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes Served user allows the presentation of his/her URI to diverted-to user = Yes 		
SIP Parameter	INVITE:		
	Request line contains ';cause=302' History-Info header: <sip:userb@networkb>;index=1, <sip: userc@networka;cause="302">;index=1.1</sip:></sip:userb@networkb>		
	181 Being Forwarded History-Info header: <sip:userb@networkb>;index=1, <sip: userc@networka;cause="408">;index=1.1 200 OK INVITE History-Info header: <sip:userb@networkb>;index=1,</sip:userb@networkb></sip:></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 C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) Communication		
	Apply post test routine		
Comments	Check: A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. Check: A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. Check: Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '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	ce/CFU		
Reference	4.5.2.6/[9]			
SELECTION EXPRESSION	SE 25			
Test purpose	Communicati	on forwarding unconditiona	al, unsuccessful NDUB.	
	Ensure that wh	d user C are in network A. The nen user A calls user B, the ca s network determined user bu	all is forwarded unconditional to use	٢
Configuration				
SIP Parameter				
Message flow				
SIP (Network A)	Int	erconnection Interface	SIP (Network B)	
		INVITE(Call-ID A-B)	→	
		CFU is performed		
	←	INVITE(Call-ID B-C)		
	<mark>486</mark>	<mark>6 Busy Here</mark> (Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	← 480	6 Busy Here(Call-ID A-B)		
		ACK(Call-ID A-B)	→	
Comments	Check: The	dialogue is terminated by rec	ceiving a 486 Busy Here.	
	Repeat this tes	st in reverse direction.		

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

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
Test purpose	options is set to presentation not allowed.
	options to set to presentation her 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:
garanen.	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]
	[4.7 22
	[any boundary name]
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ACM
	Backward call indicator
	Called party's status indicator
	no indication
	Redirection number
	Address signal (Diverted-to user)
	Call diversion information
	Notification subscription options
	presentation not allowed
	Redirecting reason
	unconditional
	Generic notification
	call is diverting
	[any boundary name]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE(Call-ID A-B)
	CFU is performed
	INVITE(Call-ID B-C, IAM)
·	183 Session Progress (Call-ID B-A, ACM)
_	Apply post test routine
Comments	Originating user in Network A establishes a call to user in Network B. Network B
	performs the diversion to a user in Network A
	Check: Is a 183 Session Progress received at the interconnection interface?
	Check: Is an ACM encapsulated in the 183?
	Check: Is the Called party's status indicator set to 'no indication'?
	Check: Is the Redirection number present?
	Check: Is Notification subscription options indicator set to 'presentation not
	allowed'?
	Check: Is the Redirecting reason set to 'unconditional'?
İ	Repeat this test in reverse direction.

Toot coop number	00 4. 010		
Test case number	SS_cfu_012 SIP-SIP/Service/CFU		
Test case group			
Reference	6.5/[24]		
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55		
Test purpose	SIP-I support. CFU performed in Network B, Notification subscription options is set to presentation allowed without redirection number.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number.		
	Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.		
Configuration	Subscription options:		
_	 Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number 		
SIP Parameter	183 Session Progress		
	Content-Type: multipart/mixed;boundary=[any boundary name][any boundary name] Content Type: application/journiversion-ity t02		
	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		
	unconditional Generic notification		
	call is diverting		
	can is diverting		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →		
	CFU is performed		
	€ INVITE(Call-ID B-C, IAM) € 183 Session Progress (Call-ID B-A, ACM)		
`	Apply post test routine		
Comments	Originating user in Network A establishes a call to user in Network B. Network B		
	performs the diversion to a user in Network A		
	Check: 183 Session Progress is received at the interconnection interface.		
	Check: Is an ACM encapsulated in the 183?		
	Check: Is the Called party's status indicator set to 'no indication'?		
	Check: Is the Redirection number present?		
	Check: Is Notification subscription options indicator set to 'presentation		
	allowed without redirection number'?		
	Check: Is the Redirecting reason set to 'unconditional'?		
	Repeat this test in reverse direction.		

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:
John San and S	 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 boundary name]
Message flow SIP (Network A)	Interconnection Interface INVITE(Call-ID A-B) CFU is performed SIP (Network B) →
	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: 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		
	number.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part		
	of Network B and is provided with CFU, Diverted-to user is subscribed to the		
	COLR service in Permanent mode.		
	Ensure that when user A calls user B, the call is forwarded unconditional to user		
	C, a Redirection number restriction parameter is present set to 'Presentation		
	restricted' in the encapsulated ANM contained in the 200 OK INVITE if		
Configuration	ISUP/BICC- SIP-I interworking is applicable in Network A. Subscription options:		
Configuration	 Connected user subscribed to COLR, Permanent = yes 		
SIP Parameter	200 OK		
on rarameter	Content-Type: multipart/mixed;boundary=[any boundary name]		
	Contone Typo: manaparemixou, boundary—[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	ANM		
	Redirection number restriction		
	Presentation restricted		
	[any boundary name]		
Message flow	[arry boundary name]		
SIP (Network A)	Interconnection Interface SIP (Network B)		
· (INVITE(Call-ID A-B), IAM →		
	CFU is performed		
•	← INVITE(Call-ID B-C)		
	180 Ringing (Call-ID C-B, ACM) →		
•	180 Ringing (Call-ID B-A)		
	200 OK INVITE (Call-ID C-B, ANM) →		
	ACK (Call-ID B-C)		
•	← 200 OK INVITE (Call-ID B-A) ACK (Call-ID A-B)		
	7 (Call 12 7 2)		
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 200 OK INVITE received at the interconnection interface?		
	Check: Is an ANM encapsulated in the 200 OK?		
	Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation		
	restricted'?		
	Repeat this test in reverse direction.		

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

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

Test case number	SC cft. 047
	SS_cfu_017
Test case group	SIP-SIP/Service/CFU
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	SIP-I support. CFU performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if
	ISUP/BICC - SIP-I interworking is applicable in Network B.
Configuration	Subscription options: Served user releases his/her number to diverted-to user = Do not release diverting numberinformation
SIP Parameter	INVITE
	Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation restricted Address signal (Diverting user) Original called number Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason unconditional
	[any boundary name]
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFU is performed
•	INVITE(Call-ID B-C, IAM) Apply post test routine
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A Check: Is a INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation restricted'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation restricted'? Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to 'unconditional'?
	Repeat this test in reverse direction.

7.1.5.6.2 Communication Forwarding Busy (CFB)

Test case number	SS_cfb_	001				
Test case group	SIP-SIP/	SIP-SIP/Service/CFB				
Reference	4.5.2.6/[9	9]				
SELECTION EXPRESSION	SE 26					
Test purpose	Commu	Communication forwarding busy, basic rules.				
		The user A and user C are in Network A. The user B is in network B and is				
		provided with CFB.				
		Ensure that when user A calls user B, the call is forwarded busy to user C. In the				
	active ca	active call state, ensure the property of speech.				
Configuration						
SIP Parameter						
Message flow						
SIP (Network A)		Interconnection Interface		SIP (Network B)		
		INVITE(Call-ID A-B)	→			
	_	CFB is performed				
	←	INVITE(Call-ID B-C)	_			
	-	180 Ringing(Call-ID C-B)	→			
	←	180 Ringing(Call-ID B-A)	_			
	-	200 OK INVITE(Call-ID C-B)	→			
	(ACK(Call-ID B-C)				
	←	200 OK INVITE(Call-ID B-A)	_			
		ACK(Call-ID A-B)	→			
		Communication				
0	011	Apply post test routine				
Comments	Check:	CDIV busy is successful.		-f -n h		
		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				
	Check:	user?	it set to the	identity of the originating		
	Repeat	his test in reverse direction.				

Test case number	SS_cfb_002			
Test case group	SIP-SIP/Service/CFB			
Reference	4.5.2.6/[9]			
SELECTION EXPRESSION	SE 26 AND SE 30			
Test purpose	Communication forwarding busy, no notification.			
	The user A and user C are in Network A. The user B is in network B and is provided with CFB, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B, the call is forwarded busy to user C, originating user is not notified.			
Configuration	Subscription options:			
	 Originating user receives notification that his communication has been diverted = No 			
SIP Parameter				
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →			
	CFB is performed ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) →			
	← 180 Ringing(Call-ID B-A) Apply post test routine			
Comments	Check: No notification regarding call forwarding in network B is received at the interconnection interface.			
	Repeat this test in reverse direction.			

Test case number	SS_cfb_003				
Test case group	SIP-SIP/Service/CFB				
Reference	4.5.2.6/[9]				
SELECTION EXPRESSION	SE 26 AND SE 30				
Test purpose	Communication forwarding busy, originating user is notified. URI from the				
	served user not received.				
	The user A and user C are in network A. The user B is in network B and is				
	provided with CFB Originating user receives notification that his communication				
	has been diverted = Yes ("Served user allows the presentation of forwarded to				
	URI to originating user in diversion notification" = No and. "Served user allows				
	the presentation of his/her URI to originating user in diversion notification" = No.				
	Ensure that when user A calls user B, the call is forwarded busy to user C, user				
	A is notified of call diversion and not informed of the diverted-to number and served user number.				
Configuration	Subscription options:				
Comigaration	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				
	<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?Privacy=history">,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)				
	← 181 Being Forwarded (Call-ID B-A)				
	180 Ringing(Call-ID C-B) →				
	← 180 Ringing(Call-ID B-A)				
0	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.				
	Tropode and tool in toroide 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:></sip:userb@networkb?reason=sip;>		
Message flow	<sip. @networka,cause="400" usero="">,index=1.1</sip.>		
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) ← 180 Ringing(Call-ID B-A)		
	Apply post test routine		
Comments	Check: A 181 Being Forwarded is received at interconnection interface. Check: A History-Info header is contained in the 181 with the URI of the diverted-to user.		
	Check: Is the cause parameter in the last entry set to '486'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.		

Test case number	SS cfb 005		
Test case group	SIP-SIP/Service/CFB		
Reference	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 26 AND SE 30		
Test purpose	Communication forwarding busy, diverted-to user does not receive the URI		
	of the served user.		
	The user A and user C are in network C. The user B is in network B and is		
	provided with CFB "Served user allows the presentation of his/her URI to the		
	diverted-to user" = No.		
	Ensure that when user A calls user B, the call is forwarded busy to user C, user		
	C is not informed of the forwarding number.		
Configuration	Subscription options:		
	• Served user allows the presentation of his/her URI to the diverted-to user =		
	No.		
SIP Parameter	INVITE:		
	Request line contains ';cause=486'		
	History-Info header:		
	<pre><sip:userb@networkb?privacy=history&reason=sip;cause=486>;index=1, <sip: userc@networka;cause="486">;index=1.1</sip:></sip:userb@networkb?privacy=history&reason=sip;cause=486></pre>		
Message flow	CSIP. USEIC @TIELWOTKA,Cause=4002,IIIuex=1.1		
SIP (Network A)	Interconnection Interface SIP (Network B)		
on (notwork //)	INVITE(Call-ID A-B) →		
	CFB is performed		
	← INVITE(Call-ID B-C)		
	Apply post test routine		
Comments	Check: A History-Info header is received in the INVITE contains the URI of		
	user B (served user) at the interconnection interface and a Privacy		
	header is escaped set to 'history'.		
	Check: Is the 'cause' parameter present in the Request line sent to user C		
	(diverted-to user) set to '486'?		
	Check: Is the cause parameter in the last entry set to '486'?		
	NOTE: The history entries can be accumulated in "one" History-Info header or		
	each history entry is present in one single History-Info header.		
	Repeat this test in reverse direction.		

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

Test case number	ICC ofb 007
Toct caco group	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>
	1.11
	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:>
	CSIP. daeto @fletwork t,oadac=4002,index=1.1
	200 OK INVITE
	200 OK INVITE
	History-Info header:
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1,</sip:userb@networkb&reason=sip;cause=486>
Message flow	History-Info header:
Message flow	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip: userc@networka;cause="486">;index=1.1</sip:></sip:userb@networkb&reason=sip;cause=486>
Message flow SIP (Network A)	History-Info header:
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip: userc@networka;cause="486">;index=1.1 Interconnection Interface</sip:></sip:userb@networkb&reason=sip;cause=486>
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface INVITE(Call-ID A-B) CFB is performed SIP (Network B)</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip: userc@networka;cause="486">;index=1.1 Interconnection Interface</sip:></sip:userb@networkb&reason=sip;cause=486>
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip: userc@networka;cause="486">;index=1.1 Interconnection Interface</sip:></sip:userb@networkb&reason=sip;cause=486>
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip: userc@networka;cause="486">;index=1.1 Interconnection Interface</sip:></sip:userb@networkb&reason=sip;cause=486>
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
SIP (Network A)	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
SIP (Network A)	History-Info header:
SIP (Network A)	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
SIP (Network A)	History-Info header:
SIP (Network A)	History-Info header:
SIP (Network A)	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
SIP (Network A)	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
SIP (Network A)	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
SIP (Network A)	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
SIP (Network A)	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>
SIP (Network A)	History-Info header: <sip:userb@networkb&reason=sip;cause=486>;index=1, <sip:userc@networka;cause=486>;index=1.1 Interconnection Interface</sip:userc@networka;cause=486></sip:userb@networkb&reason=sip;cause=486>

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

Test case number	SS_cfb_0	09		
Test case group		Service/CFB		
Reference	4.5.2.6/[9]			
SELECTION EXPRESSION				
	SE 26			
Test purpose	Commun	ication forwarding busy, unsue	ccessful N	DUB.
	The user	A and user C are in network A. T	he user B is	s in network B and is
	provided v	with CFB.		
	Ensure th	at when user A calls user B, the	call is forwa	rded busy to user C and
	user C is	network determined user busy.		•
Configuration				
SIP Parameter				
Message flow	1			
SIP (Network A)		Interconnection Interface		SIP (Network B)
(10111011114)		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		
	Check:	The dialogue is terminated by re	eceiving a 4	·86 Busy Here.
	Repeat th	is test in reverse direction.		

Test case number	SS_cfb_010	
Test case group	SIP-SIP/Service/CFB	
Reference	4.5.2.6/[9]	
SELECTION EXPRESSION	SE 26 AND SE 30 AND [Network A] SE 9	
Test purpose	Communication forwarding busy, interaction with a not trusted network.	
	The user A and user C are in network A. The user B is in network B and is provided with CFB Originating user receives notification that his communication has been diverted = Yes ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting number is released to the diverted-to user"=Yes. Ensure that when user A calls user B, the call is forwarded busy to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.	
Configuration	Subscription options:	
	 Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to the diverted-to user = No 	
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed INVITE(Call-ID B-C) 181 Being Forwarded (Call-ID B-A) Apply post test routine	
Comments	Check: No History-Info header is received in the INVITE at the interconnection	
	interface. Check: No History-Info header is received in the 181 Being Forwarded at the interconnection interface (if sent). Repeat this test in reverse direction.	

Test case number Test case group SIP-SIP/Service/CFB Reference 6.5/[24] SELECTION EXPRESSION 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 of Network B and is provided with CFB, Calling user receives notification to call has been diverted (forwarded or deflected) = yes, without diverted-to unumber. Ensure that when user A calls user B, the call is forwarded on busy user to C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained 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 (forward or deflected) = no SIP Parameter 183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required	I part hat his user
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 of Network B and is provided with CFB, Calling user receives notification to call has been diverted (forwarded or deflected) = yes, without diverted-to unumber. Ensure that when user A calls user B, the call is forwarded on busy user to C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B. Subscription options: Calling user receives notification that his call has been diverted (forward of deflected) = no	I part hat his user
SELECTION EXPRESSION [Network B] SE 17 AND SE 47 AND SE 55	I part hat his user
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 of Network B and is provided with CFB, Calling user receives notification to call has been diverted (forwarded or deflected) = yes, without diverted-to unumber. Ensure that when user A calls user B, the call is forwarded on busy user to C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained 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 (forward or deflected) = no 183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required	I part hat his user o user
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 of Network B and is provided with CFB, Calling user receives notification to call has been diverted (forwarded or deflected) = yes, without diverted-to unumber. Ensure that when user A calls user B, the call is forwarded on busy user to C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained 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 (forwator 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	I part hat his user o user
of Network B and is provided with CFB, Calling user receives notification the call has been diverted (forwarded or deflected) = yes, without diverted to unumber. Ensure that when user A calls user B, the call is forwarded on busy user to C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained 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 (forwator 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	hat his user user
C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained 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 (forwator 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	
Configuration Subscription options: Calling user receives notification that his call has been diverted (forward or deflected) = no SIP Parameter 183 Session Progress	
Calling user receives notification that his call has been diverted (forward 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	
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	ırded
Content-Type: multipart/mixed;boundary=[any boundary name][any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required	
Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required	
Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required	
Content-Disposition: signal;handling=required	
ACM	
I 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	
User Busy	
Generic notification	
call is diverting	
[any boundary name]	
Message flow	
SIP (Network A) Interconnection Interface SIP (Network INVITE(Call-ID A-B)	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. Netv	vork B
performs the diversion to a user in Network A	
Check: Is a 183 Session Progress received at the interconnection interf	ace?
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'?	
Check: Is the Redirecting reason set to User Busy'?	not
Repeat this test in reverse direction.	not

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

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

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

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, nandling=required		
	ANM		
	Redirection number restriction		
	Presentation allowed		
	Or		
	Redirection number restriction not present		
	Troumbon Tourist Processing		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
, , ,	INVITE(Call-ID A-B), IAM →		
	CFB is performed		
•	F INVITE(Call-ID B-C)		
	180 Ringing (Call-ID C-B, ACM) →		
•	180 Ringing (Call-ID B-A)		
	200 OK INVITE (Call-ID C-B, ANM) →		
	ACK (Call-ID B-C)		
•	£ 200 OK INVITE (Call-ID B-A)		
	ACK (Call-ID A-B) →		
	Apply post test routine		
Comments	Originating user in Network A establishes a call to user in Network B. Network B		
	performs the diversion to a user in Network A		
	Check: Is a 200 OK INVITE received at the interconnection interface?		
	Check: Is an ANM encapsulated in the 200 OK?		
	Check: Is the ISUP/BICC Redirection number restriction present set to		
	'Presentation allowed' or is the parameter absent?		
	Repeat this test in reverse direction.		

Toot coop number	CC -th 040		
Test case number	SS_cfb_016		
Test case group	SIP-SIP/Service/CFB		
Reference	7.1/[24]		
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55		
Test purpose	SIP-I support. CFB performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded on busy user to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.		
Configuration			
Configuration	 Subscription options: Served user releases his/her number to diverted-to user = Release diverting number information 		
SIP Parameter	INVITE		
	Content-Type: multipart/mixed;boundary=[any boundary name][any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	IAM		
	Redirecting number		
	Address presentation restricted indicator presentation allowed		
	Address signal (<i>Diverting user</i>)		
	Original called number Address presentation restricted indicator presentation allowed		
	Address signal		
	Redirection information Original Redirection Reason		
	unknown		
	Redirecting indicator		
	Redirection counter		
	Redirecting reason		
	User Busy		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B) →		
	CFB is performed		
│	<mark>F INVITE</mark> (Call-ID B-C, <mark>IAM</mark>)		
	Apply post test routine		
Comments	Originating user in Network A establishes a call to user in Network B. Network B		
	performs the diversion to a user in Network A		
	Check: Is a INVITE request received at the interconnection interface?		
	Check: Is an IAM encapsulated in the INVITE?		
	Check: Is the Redirecting number present and the Address presentation		
	restricted indicator is set to 'presentation allowed'?		
	Check: Is the Original called number present and the Address presentation		
	restricted indicator is set to 'presentation allowed'?		
	Check: Is the Redirection number present?		
	Check: Is Redirection information present and the Redirecting reason is set to		
	'User Busy'?		
	Repeat this test in reverse direction.		
	To the same and a second secon		

Test case number SS_cfb Test case group SIP-SI	
	P/Service/CFB
Reference 7.1/[24	
SELECTION EXPRESSION [Netwo	ork B] SE 17 AND SE 47 AND SE 55
Redire	support. CFB performed in Network B, Notification of diverted-to user ecting number 'presentation restricted'.
of Netv diverte Ensure C, use The no Redire	ser A and user C are in Network A. The user B is in the PSTN/PLMN part work B and is provided with CFB, Served user releases his/her number to ed-to user = Release diverting number information. The that when user A calls user B, the call is forwarded on busy user to user a contified of call diversion and informed of the diverting number. The provided in the continuous presentation restricted and Redirection information if
	BICC - SIP-I interworking is applicable in Network B.
• Se	ription options: erved user releases his/her number to diverted-to user = Do not release verting numberinformation
SIP Parameter INVITE	
	Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation restricted Address signal (Diverting user) Original called number Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason 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 Origina	ating user in Network A establishes a call to user in Network B. Network B
	Insight the diversion to a user in Network A Is a INVITE request received at the interconnection interface? Is an IAM encapsulated in the INVITE? Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation restricted'? Is the Original called number present and the Address presentation restricted indicator is set to 'presentation restricted'? Is the Redirection number present?
Repea	t 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	Commu	nication forwarding no reply, ba	sic rules.	
	The user	A and user C are in Network A. T	he user B i	s in network B and is
		with CFNR.		
		hat 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	7	
		• • • • • • • • • • • • • • • • • • • •		
Comments	Chaola	Apply post test routine		
Comments	Check: Check:	CDIV no reply is successful.	o proporty	of another
	Check:	In the active call state, ensure the Is the P-Asserted-Identity preser		
	Check:	user?	ii sei io ine	identity of the originating
	Reneat t	his test in reverse direction.		
	Nepear	no toot in reverse unection.		

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	International Conductories OID (ALC) D)		
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →		
	=(•«		
	180 Ringing(Call-ID B-A) CFB is performed		
	← INVITE(Call-ID B-C)		
	181 Being Forwarded (Call-ID B-A)		
	180 Ringing(Call-ID C-B) →		
	← 180 Ringing(Call-ID B-A)		
	Apply post test routine		
Comments	Check: A 181 Being Forwarded and a History-Info header is received at the		
	interconnection interface in both entries in the History-Info header a		
	Privacy header is escaped value 'history'.		
	Check: Is the cause parameter in the last entry is set to '408'?		
	NOTE: The history entries can be accumulated in "one" History-Info header or		
	each history entry is present in one single History-Info header.		
	Repeat this test in reverse direction.		

Test case number	SS_cfnr_004		
Test case group	SIP-SIP/Service/CFNR		
Reference	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 27 AND SE 30		
Test purpose	Communication forwarding no reply, originating user is notified. URI from the diverted-to user received.		
	The user A and user C are in network A. The user B is in network B and is provided with CFNR Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes.		
	Ensure that when user A calls user B, the call is forwarded no reply to user C,		
	user A is notified of call diversion and informed of the diverted-to number.		
Configuration	 Subscription options: Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes 		
SIP Parameter	181 Being Forwarded		
on rarameter	<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) ← 181 Being Forwarded (Call-ID B-A) 180 Ringing (Call-ID C-B) 		
	← 180 Ringing(Call-ID B-A) Apply post test routine		
Comments	Check: A 181 Being Forwarded is received at the interconnection interface. Check: A History-Info header is contained in the 181 with the URI of the diverted-to user. Check: Is the cause parameter in the last entry is set to '408'?		
	NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.		

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

Test case number	CC ofor OOC		
	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 soos number	00 star 007		
Test case number	SS_cfnr_007		
Test case group	SIP-SIP/Service/CFNR		
Reference	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 27 AND SE 30		
Test purpose	Communication forwarding no reply, full notification.		
	The user A and user C are in network A. The user B is in network B and is		
	provided with CFNR Originating user receives notification that his		
	communication has been diverted = Yes, ("Served user allows the presentation		
	of forwarded to URI to originating user in diversion notification" = Yes, "diverting		
	number is released to the diverted-to user" = Yes.		
	Ensure that when user A calls user B, the call is forwarded no reply to user C,		
	user A is notified of call diversion and informed of the diverted-to number and		
	user C is informed of the forwarding number.		
Configuration	Subscription options:		
	 Originating user receives notification that his communication has been 		
	diverted = Yes		
	 Served user allows the presentation of forwarded to URI to originating user 		
	in diversion notification = Yes		
	 diverting number is released to the diverted-to user = Yes 		
SIP Parameter	INVITE:		
	Request line contains ';cause=408'		
	History-Info header:		
	<pre><sip:userb@networkb&reason=sip;cause=408>;index=1,</sip:userb@networkb&reason=sip;cause=408></pre>		
	<sip: userc@networka;cause="486">;index=1.1</sip:>		
	101p. 40010 @110tW01W1,04400=1002,11140X=1.11		
	181 Being Forwarded		
	History-Info header:		
	<sip:userb@network>;index=1,</sip:userb@network>		
	<sip: userc@networka;cause="408">;index=1.1</sip:>		
	solp. addite characteristics and additional specific 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 by the control of		
SIP (Network A)	Interconnection Interface SIP (Network B)		
J. (1.5.11.7.)	INVITE(Call-ID A-B) →		
	← 180 Ringing(Call-ID B-A)		
	CFB is performed		
	← INVITE(Call-ID B-C)		
	← 181 Being Forwarded (Call-ID B-A		
	180 Ringing(Call-ID C-B) →		
	← 180 Ringing(Call-ID B-A)		
	200 OK INVITE(Call-ID C-B) →		
	← ACK(Call-ID C-B)		
	← 200 OK INVITE(Call-ID B-A)		
	ACK(Call-ID A-B) →		
	Apply post test routine		
Comments	Check: A History-Info header is received in the INVITE at the interconnection		
Comments	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.		
	the diverted-to user. Check: Is the 'cause' parameter present in the Request line sent to user C		
	the diverted-to user. Check: Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '408'?		
	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'?		
	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		
	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	·		
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.		

To at a see a secondo a se	100 (000		
Test case number	SS_cfnr_009			
Test case group	SIP-SIP/Service/CFNR			
Reference	4.5.2.6/[9)]		
SELECTION EXPRESSION	SE 27			
Test purpose	Communication forwarding no reply, unsuccessful NDUB.			
	provided Ensure th	A and user C are in network A. T with CFNR. nat when user A calls user B, the C is network determined user bus	call is forwa	
Configuration				
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface INVITE(Call-ID A-B)	→	SIP (Network B)
	←	180 Ringing(Call-ID B-A) CFB is performed	-	
	←	INVITE(Call-ID B-C) 486 Busy Here(Call-ID C-B)	→	
	(ACK(Call-ÌD 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	86 Busy Here.
	Repeat th	nis test in reverse direction.		

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

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

Test case number	CC ofor 012			
	SS_cfnr_012 SIP-SIP/Service/CFNR			
Test case group Reference				
SELECTION EXPRESSION	6.5/[24] [Network B] SE 17 AND SE 47 AND SE 55			
	SIP-I support. CFNR performed in Network B, Notification subscription			
Test purpose	options is set to presentation allowed without redirection number.			
	options is set to presentation anowed 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			
	000			
	CPG			
	Event indicator			
	Alerting or Progress Redirection number			
	Address signal (<i>Diverted-to user</i>)			
	Call diversion information			
	Notification subscription options			
	presentation allowed without redirection number			
	Redirecting reason			
	No reply			
	Generic notification			
	call is diverting			
	, and the second			
	[any boundary name]			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE(Call-ID A-B) →			
·	180 Ringing (Call-ID B-A, ACM)			
	CFNR is performed			
	► INVITE(Call-ID B-C, IAM) ► 183 Session Progress (Call-ID B-A, ACM)			
	183 Session Progress (Call-ID B-A, ACM) Apply post test routine			
Comments	Originating user in Network A establishes a call to user in Network B. Network B			
	performs the diversion to a user in Network A			
	Check: 183 Session Progress is received at the interconnection interface.			
	Check: Is an CPG encapsulated in the 183?			
	Check: is the Called party's status indicator set to 'no indication'?			
	Check: Is the Redirection number present?			
	Check: Is Notification subscription options indicator is set to 'presentation			
	allowed without redirection number'?			
	Check: Is the Redirecting reason set to 'No reply'?			
	Repeat this test in reverse direction.			

Test case number	SS_cfnr_013			
	SIP-SIP/Service/CFNR			
Test case group Reference	6.5/[24]			
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55			
	SIP-I support. CFNR performed in Network B, Notification subscription			
Test purpose	options is set to presentation 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:			
Comiguration	 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			
Massage flow	[any boundary name]			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)			
on (Network A)	INVITE(Call-ID A-B)			
	← 180 Ringing (Call-ID B-A, ACM)			
	CFNR is performed			
	← INVITE(Call-ID B-C, IAM)			
	← 183 Session Progress (Call-ID B-A, ACM)			
	Apply post test routine			
Comments	Originating user in Network A establishes a call to user in Network B. Network B			
	performs the diversion to a user in Network A			
	Check: 183 Session Progress is received at the interconnection interface.			
	Check: Is an CPG encapsulated in the 183?			
	Check: Is the Called party's status indicator set to 'no indication'?			
	Check: Is the Redirection number present?			
	Check: Is Notification subscription options indicator is set to 'presentation allowed with redirection number'?			
	Check: Is the Redirecting reason set to 'No reply'?			
	Repeat this test in reverse direction.			
	proposit and tool in reverse unconorn.			

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 hayandany nama]		
	[any boundary name] Content-Type: application/isup;version=itu-t92		
	Content-Type: application/isup,version=itu-t92 Content-Disposition: signal;handling=required		
	Content-Disposition. signal, nanding=required		
	ANM		
	Redirection number restriction		
	Presentation restricted		
	1 1000 matter 100th old		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B), IAM →		
	← 180 Ringing (Call-ID B-A, ACM)		
	CFNR is performed		
	← INVITE(Call-ID B-C)		
	180 Ringing (Call-ID C-B, ACM) →		
	← 180 Ringing (Call-ID B-A)		
	200 OK INVITE (Call-ID C-B, ANM) →		
	ACK (Call-ID B-C)		
	200 OK INVITE (Call-ID B-A)		
	ACK (Call-ID A-B) →		
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 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.		
	Incheat tills test ill 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 ISUP/BICC- SIP-I interworking is applicable in Network A.			
Configuration	Subscription options:			
- Comigaration	 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			
	Redirection number restriction Presentation allowed			
	or			
	Redirection number restriction not present			
	[any havedow none]			
1	[any boundary name]			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B), IAM →			
•	180 Ringing (Call-ID B-A) CFNR is performed			
•	INVITE(Call-ID B-C)			
	180 Ringing (Call-ID C-B, ACM) →			
	180 Ringing (Call-ID B-A)			
	200 OK INVITE (Call-ID C-B, ANM) →			
	ACK (Call-ID B-C)			
•	200 OK INVITE (Call-ID B-A)			
	ACK (Call-ID A-B) →			
	Apply post test routine			
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A			
	Check: Is a 200 OK INVITE received at the interconnection interface?			
	Check: Is an ANM encapsulated in the 200 OK?			
	Check: Is the ISUP/BICC Redirection number restriction present set to			
	'Presentation allowed' or is the parameter absent?			
	Repeat this test in reverse direction.			

Test case number	SS_cfnr_016		
Test case group	SIP-SIP/Service/CFNR		
Reference	7.1/[24]		
SELECTION EXPRESSION			
Test purpose	SIP-I support. CFNR performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded on no reply to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if		
	ISUP/BICC - SIP-I interworking is applicable in Network B.		
Configuration	Subscription options:		
	Served user releases his/her number to diverted-to user = Release diverting		
OID Developed	number information		
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
Message flow SIP (Network A)	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 No reply [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)		
	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			
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 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:			
garanon	 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 indicator			
	Address signal (<i>Diverting user</i>)			
	Original called number			
	Address presentation restricted indicator			
	presentation restricted			
	Address signal			
	Redirection information			
	Original Redirection Reason			
	unknown			
	Redirecting indicator			
	Redirection counter			
	Redirecting reason			
	No reply			
Managa flaw	[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)			
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 restricted'?			
	Check: Is the Original called number present and the Address presentation			
	restricted indicator is set to 'presentation restricted'?			
	Check: Is the Redirection number present?			
	Check: Is Redirection information present and the Redirecting reason is set to			
	'No reply'?			
<u> </u>	Repeat this test in reverse direction.			

7.1.5.6.4 Communication Forwarding Not Logged in (CFNL)

Test case number	SS_cfnl_			
Test case group	SIP-SIP/	SIP-SIP/Service/CFNL		
Reference	4.5.2.6/[9	9]		
SELECTION EXPRESSION	SE 28			
Test purpose	Commu	nication forwarding not logged i	in, basic rι	iles.
	provided	A and user C are in Network A. T with CFNL.		
	Ensure th	nat when user A calls user B, the o	call is forwa	rded not logged in 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) 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)	→ →	SIP (Network 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: Check:	The CDIV not logged in is succe. In the active call state, ensure the Is the P-Asserted-Identity preser user? his test in reverse direction.	e property	

Test case number	SS_cfnl_	002		
Test case group	SIP-SIP/	SIP-SIP/Service/CFNL		
Reference	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 28 AN	ND SE 30		
Test purpose	Commu	nication forwarding not logged	in, no notif	ication.
	provided that his c Ensure th	A and user C are in Network A. with CFNL, subscription option: communication has been diverted nat when user A calls user B, the ating user is not notified.	Originating t	user receives notification
Configuration	Subscrip	otion options:		
	 Original 	nating user receives notification	that his com	munication has been
	dive	ted = No		
SIP Parameter				
Message flow SIP (Network A)	·	Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CFNL is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		Apply post test routine		
Comments	Check:	No notification regarding call fo	rwarding in r	network B is received at
		interconnection interface.	-	
	Repeat th	nis test in reverse direction.		

SELECTION EXPRESSION SE 28 AND SE 30 Communication forwarding not logged in, originating user is notified. URI of the diverted-to user not received. The user A and user C are in network A. The user B is in network B and is provided with CFNL Originating user receives notification that his communication has been diverted = Yes and ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and not informed of the diverted-to number and the served user number. Configuration Subscription options: Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to originating user in diversion notification = No SIP Parameter 181 Being Forwarded	Test case number	SS_cfnl_003			
SELECTION EXPRESSION SE 28 AND SE 30	Test case group	SIP-SIP/Service/CFNL			
Test purpose Communication forwarding not logged in, originating user is notified. URI of the diverted-to user not received. The user A and user C are in network A. The user B is in network B and is provided with CFNL Originating user receives notification that his communication has been diverted = Yes and ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and not informed of the diverted-to number and the served user number. 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 originating user in diversion notification = No SiP Parameter is 18 tesing Forwarded	Reference	4.5.2.6/[9]			
The user A and user C are in network A. The user B is in network B and is provided with CFNL Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and not informed of the diverted-to number and the served user number. Configuration Subscription options: Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to originating user in diversion notification = No SIP Parameter 181 Being Forwarded SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) 180 Ringing(Call-ID B-A) Apply post test routine Comments Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value "history". Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entry is present in one single History-Info header.	SELECTION EXPRESSION				
The user A and user C are in network A. The user B is in network B and is provided with CFNL Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and not informed of the diverted-to number and the served user number. Configuration Subscription options: Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to originating user in diversion notification = No SIP Parameter 181 Being Forwarded SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) 180 Ringing(Call-ID B-A) Apply post test routine Comments Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value "history". Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entry is present in one single History-Info header.	Test purpose				
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 Served user allows the presentation of his/her URI to originating user in diversion notification = No SIP Parameter Ist Being Forwarded <isip:userb@networkb?privacy=history>;index=1, <isip:userc@networka;cause=404?privacy=history>;index=1.1 Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) Ist Being Forwarded(Call-ID B-A) Apply post test routine Check: A 181 Being Forwarded and a History-Info header is received at 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.</isip:userc@networka;cause=404?privacy=history></isip:userb@networkb?privacy=history>					
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 Served user allows the presentation of his/her URI to originating user in diversion notification = No SIP Parameter 181 Being Forwarded <isip:usert@enetworkb?privacy=history>:index=1. <isip:usert@enetworka?cause=404?privacy=history>:index=1.1 Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) Interconnection Interface INVITE(Call-ID B-A) Apply post test routine 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.</isip:usert@enetworka?cause=404?privacy=history></isip:usert@enetworkb?privacy=history>					
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 Isl Being Forwarded <ii>isip:userB@networkB?Privacy=history>;index=1, <iijp:userb@networkb?privacy=history>;index=1.1 Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) Isl Being Forwarded(Call-ID B-A) 180 Ringing(Call-ID B-C) Isl Being Forwarded(Call-ID B-A) Apply post test routine 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.</iijp:userb@networkb?privacy=history></ii>					
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 <pre> <ipre> </ipre></pre> <pre> <ipre> </ipre></pre> <pre> </pre> <pre></pre>					
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 SIP Parameter 181 Being Forwarded <pre> <ipre> <ipre> </ipre></ipre></pre> <pre> <ipre> </ipre></pre> <pre></pre>					
Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and not informed of the diverted-to number and the served user number. Configuration Subscription options: Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to originating user in diversion notification = No SIP Parameter 181 Being Forwarded SIP (Network B) = No 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) INVITE(Call-ID B-A) INVITE(C					
C, user A is notified of call diversion and not informed of the diverted-to number and the served user number. Subscription options: Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to originating user in diversion notification = No SIP Parameter 181 Being Forwarded <sip: ?="" @="" network="" privacy="history" user="">: index=1, <sip: ?="" @="" network="" privacy="history" user="">: index=1.1 Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) 181 Being Forwarded (Call-ID B-A) 180 Ringing(Call-ID C-B) 180 Ringing(Call-ID C-B) 180 Ringing(Call-ID B-A) Apply post test routine Comments Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header.</sip:></sip:>					
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 Ist Being Forwarded					
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: user@networkb?privacy="history">;index=1, <sip: user@networkb?privacy="history">;index=1.1 Message flow SIP (Network A) Interconnection Interface SIP (Network B)</sip:></sip:>		· ·			
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,	Configuration				
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 <pre></pre>	Comiguration				
• 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		diverted – Vas			
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 <pre></pre>					
Served user allows the presentation of his/her URI to originating user in diversion notification = No SIP Parameter 181 Being Forwarded <sip:userb@networkb?privacy=history>;index=1, <sip:userc@networka;cause=404?privacy=history>;index=1.1 Message flow SIP (Network A) Interconnection Interface 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 Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.</sip:userc@networka;cause=404?privacy=history></sip:userb@networkb?privacy=history>					
diversion notification = No SIP Parameter 181 Being Forwarded <pre></pre>					
SIP Parameter					
<pre></pre>	SIP Parameter				
Sip: userC @networkA;cause=404?Privacy=history>;index=1.1 Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFNL is performed INVITE(Call-ID B-C) 181 Being Forwarded (Call-ID B-A) 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID B-A) Apply post test routine Comments Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.					
SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID B-C) INVITE(Call-ID B-A) INVITE(Call-ID B-C) INVITE(Call-ID B-A) 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 B-A) INVITE(C		<pre><sip: userc@networka;cause="404?Privacy=history">;index=1.1</sip:></pre>			
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) Apply post test routine Comments Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.	Message flow				
CFNL is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded (Call-ID B-A) 180 Ringing(Call-ID C-B) ← 180 Ringing(Call-ID B-A) Apply post test routine Comments Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.	SIP (Network A)				
 ★ INVITE(Call-ID B-C) ★ 181 Being Forwarded (Call-ID B-A) 180 Ringing(Call-ID C-B) ★ 180 Ringing(Call-ID B-A) Apply post test routine Comments Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. 		(/			
 ★ 181 Being Forwarded (Call-ID B-A)					
Table Ringing(Call-ID C-B) 180 Ringing(Call-ID B-A) Apply post test routine Comments Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.					
Table Ringing (Call-ID B-A) Apply post test routine Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.		- 10.120.11g 10.11a.1a.a.(0a.11 12.27.1)			
Comments Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.					
Check: A 181 Being Forwarded and a History-Info header is received at theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.					
theinterconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.	Comments				
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.					
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.					
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.					
		Repeat this test in reverse direction.			

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

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

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

Test case number	SS cfnl 007
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	Communication forwarding not logged in, full notification.
Configuration	The user A and user C are in network A. The user B is in network B and is provided with CFNL Originating user receives notification that his communication has been diverted = Yes, ("Served user allows the presentation of forwarded to URI to originating user in diversion notification" =Yes, "diverting number is released to the diverted-to user" =Yes. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number. Subscription options:
	 Originating user receives notification that his communication has been
	 diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes diverting number is released to the diverted-to user = Yes
SIP Parameter	INVITE:
	Request line contains ';cause=404' History-Info header: <sip:userb@networkb&reason=sip;cause=404< a="">;index=1, <sip: userc@networka;cause="404">;index=1.1</sip:></sip:userb@networkb&reason=sip;cause=404<>
	191 Boing Forwarded
	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:userb@networkb></sip:></sip:userb@network>
	<sip: userc@networka;cause="404">;index=1.1</sip:>
Message flow 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 C-B) ACK(Call-ID C-B) COUNTY (Call-ID C-B) ACK(Call-ID C-B) ACK(Call-ID A-B) ACK(Call-ID A-B) APPly post test routine
Comments	 Check: A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. Check: A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. Check: Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '404'? Check: Is the cause parameter in the last entry is set to '404'? NOTE: The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. Repeat this test in reverse direction.

Test case number	SS cfnl 008
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[9]
SELECTION EXPRESSION	SE 28
Test purpose	Communication forwarding not logged in, unsuccessful UDUB.
	The user A and user C are in network A. The user B is in network B and is
	provided with CFNL.
	Ensure that when user A calls user B, the call is forwarded not logged in to user
	C and user C is user determined user busy.
Configuration	
SIP Parameter	
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
, , ,	INVITE(Call-ID A-B) →
	CFNL is performed
	486 Busy Here (Call-ID C-B) →
	ACK(Call-ID B-C)
	486 Busy Here(Call-ID A-B)
	ACK(Call-ID A-B) →
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here.
Comments	
	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.

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 use C, user A is notified of call diversion and not informed of the diverted-to numbe 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 use 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 in diversion notification = No INVITE: no History-Info header INVITE: no History-Info header INVITE: no History-Info header INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) Till 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.	Test case number	SS_cfnl_010
Reference 4.5.2.6/[9] SELECTION EXPRESSION SE 28 AND SE 30 AND [Network A] SE 9	Test case group	SIP-SIP/Service/CFNL
Test purpose Communication forwarding not logged in, interaction with a not trusted network. The user A and user C are in network A. The user B is in network B and is provided with CFNL Originating user receives notification that his communication has been diverted = Yes ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting number is released to the diverted-to user"=Yes. Ensure that when user A calls user B, the call is forwarded not logged in to use C, user A is notified of call diversion and not informed of the diverted-to numbe and user C is not informed of the forwarding number. Configuration 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 use 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 in No Served user allows the presentation of his/her URI to the diverted-to user in No No Interconnection Interface INVITE (Call-ID B-B) CFNL is performed INVITE(Call-ID B-B) CFNL is performed INVITE(Call-ID B-C) (IBI 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 interface.		4.5.2.6/[9]
Test purpose Communication forwarding not logged in, interaction with a not trusted network.	SELECTION EXPRESSION	
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 use C, user A is notified of call diversion and not informed of the diverted-to numbe 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 in diversion notification = No INVITE: no History-Info header INVITE: no History-Info header INVITE: no History-Info header INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) Total Being Forwarded (Call-ID B-A) Apply post test routine 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. Check: No History-Info header is received in the 181 Being Forwarded at the		Communication forwarding not logged in, interaction with a not trusted
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 in No No Served user allows the presentation of his/her URI to the diverted-to user in No No INVITE: no History-Info header 181 Being Forwarded no History-Info header Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFNL is performed No 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		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.
Subscription options: Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating use 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		C, user A is notified of call diversion and not informed of the diverted-to number
in diversion notification = No Served user allows the presentation of his/her URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to the diverted-to user = No INVITE: no History-Info header 181 Being Forwarded no History-Info header Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) **INVITE(Call-ID B-C) **INVITE(Call-ID B-A) Apply post test routine Comments Check: No History-Info header is received in the INVITE at the interconnection interface. Check: No History-Info header is received in the 181 Being Forwarded at the	Configuration	 Subscription options: Originating user receives notification that his communication has been diverted = Yes
SIP Parameter INVITE: no History-Info header 181 Being Forwarded no History-Info header		 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 =
Message flow 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) 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.	SIP Parameter	INVITE: no History-Info header
INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) I81 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 content of the interconnection interface.	Message flow	,
interface. Check: No History-Info header is received in the 181 Being Forwarded at the	SIP (Network A)	INVITE(Call-ID A-B) → CFNL is performed INVITE(Call-ID B-C) 181 Being Forwarded(Call-ID B-A) Apply post test routine
Repeat this test in reverse direction.	Comments	interface. Check: No History-Info header is received in the 181 Being Forwarded at the interconnection interface (if sent).

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
Configuration	interworking is applicable in Network B. Subscription options:
Configuration	
	 Calling user receives notification that his call has been diverted (forwarded or deflected) = no
SIP Parameter	183 Session Progress
	Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name]
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ACM
	Backward call indicator
	Called party's status indicator
	no indication
	Redirection number
	Address signal (Diverted-to user)
	Call diversion information
	Notification subscription options
	presentation not allowed
	Redirecting reason
	Mobile subscriber not reachable
	Generic notification
	call is diverting
Message flow	[any boundary name]
SIP (Network A)	Interconnection Interface SIP (Network B)
On (NOTWORK A)	INVITE(Call-ID A-B)
	CFNL is performed
	INVITE(Call-ID B-C, IAM)
	183 Session Progress (Call-ID B-A, ACM)
	Apply post test routine
Comments	Originating user in Network A establishes a call to user in Network B. Network B
	performs the diversion to a user in Network A
	Check: Is a 183 Session Progress received at the interconnection interface?
	Check: Is an ACM encapsulated in the 183?
	Check: Is the Called party's status indicator set to 'no indication'?
	Check: Is the Redirection number present?
	Check: Is Notification subscription options indicator set to 'presentation not allowed'?
	Check: Is the Redirecting reason set to 'Mobile subscriber not reachable'?
	Repeat this test in reverse direction.
	proposition toot in reverse direction.

Toot ooo weeks	CC -f-1 040
Test case number	SS_cfnl_012
Test case group	SIP-SIP/Service/CFNL
Reference	6.5/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	SIP-I support. CFNL performed in Network B, Notification subscription options is set to presentation allowed without redirection number.
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number.
	Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC
	interworking is applicable in Network B.
Configuration	Subscription options:
	 Calling user receives notification that his call has been diverted (forwarded or deflected) = 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 Mobile subscriber not reachable
	Generic notification
	call is diverting
	[any boundary name]
Message flow	· · · · · · · · · · · · · · · · · · ·
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →
	CFNL is performed
	INVITE(Call-ID B-C, IAM)
·	← 183 Session Progress (Call-ID B-A, ACM)
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: 183 Session Progress is received at the interconnection interface.
	Check: Is an ACM encapsulated in the 183?
	Check: Is the Called party's status indicator set to 'no indication'?
	Check: Is the Redirection number present?
	Check: Is Notification subscription options indicator is set to 'presentation
	allowed without redirection number'?
	Check: Is the Redirecting reason set to 'Mobile subscriber not reachable'?
	Repeat this test in reverse direction.
-	

Test case number	SS_cfnl_013			
Test case group	SIP-SIP/Service/CFNL			
Reference	6.5/[24]			
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55			
Test purpose	SIP-I support. CFNL performed in Network B, Notification subscription options is set to presentation allowed with redirection number.			
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNL, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number.			
	Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user A is notified of call diversion and informed of the diverted-to number.			
	The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.			
Configuration	Subscription options:			
	 Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number 			
SIP Parameter	183 Session Progress			
	Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name]			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	ACM			
	Backward call indicator			
	Called party's status indicator no indication			
	Redirection number			
	Address signal (<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			
	[any boundary name]			
Message flow	j [any boundary name]			
SIP (Network A)	Interconnection Interface SIP (Network B)			
.	INVITE(Call-ID A-B) →			
	CFNL is performed			
	INVITE(Call-ID B-C, IAM)			
 	183 Session Progress (Call-ID B-A, ACM)			
Comments	Apply post test routine			
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 havedow rane]				
	[any boundary name] Content-Type: application/isup;version=itu-t92				
	Content-Disposition: signal;handling=required				
	ANM				
	Redirection number restriction Presentation restricted				
	[any boundary name]				
Message flow					
SIP (Network A)	Interconnection Interface SIP (Network B)				
	INVITE(Call-ID A-B), IAM →				
	CFNL is performed				
	← INVITE(Call-ID B-C)				
	180 Ringing (Call-ID C-B, ACM) →				
	← 180 Ringing (Call-ID B-A)				
	200 OK INVITE (Call-ID C-B, ANM) →				
	ACK (Call-ID B-C)				
	← 200 OK INVITE (Call-ID B-A) ACK (Call-ID A-B)				
	/ (ean 12 / (2)				
Commonts	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.				
	Interpolation to the foreign 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 pa			
	of Network B and is provided with CFNL, Diverted-to user is not subscribed to			
	the COLR service.			
	Ensure that when user A calls user B, the call is forwarded not logged in to user			
	C, if a Redirection number restriction parameter is present it is set to			
	'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE			
	if ISUP/BICC- SIP-I interworking is applicable in Network A.			
Configuration	Subscription options:			
	Connected user subscribed to COLR = no			
SIP Parameter	200 OK			
	Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any houndary name]			
	[any boundary name] Content-Type: application/isup;version=itu-t92			
	Content-Type: application/isup,version=itu-is2 Content-Disposition: signal;handling=required			
	Content-Disposition: Signal, nationing-required			
	ANM			
	Redirection number restriction			
	Presentation allowed or Redirection number restriction not present			
	[any boundary name]			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE(Call-ID A-B), IAM →			
	CFNL is performed			
	← INVITE(Call-ID B-C)			
	180 Ringing (Call-ID C-B, ACM) →			
	← 180 Ringing (Call-ID B-A) 200 OK INVITE (Call-ID C-B. ANM)			
	← ACK (Call-ID B-C) ← 200 OK INVITE (Call-ID B-A)			
	← 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			
	li di			
	•			
	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 a 200 OK INVITE received at the interconnection interface?			

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:				
Configuration	 Served user releases his/her number to diverted-to user = Release diverting 				
	number information				
SIP Parameter	INVITE				
on randicier	Content-Type: multipart/mixed;boundary=[any boundary name]				
	[any boundary name]				
	Content-Type: application/isup;version=itu-t92				
	Content-Disposition: signal;handling=required				
	IAM Source of the second secon				
	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	[arry bourtacity flattlo]				
SIP (Network A)	Interconnection Interface SIP (Network B)				
Cii (iiciiiciii)	INVITE(Call-ID A-B)				
	CFNL is performed				
•	NVITE(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				
	'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:			
Comiguration	 Served user releases his/her number to diverted-to user = Do not release 			
	diverting numberinformation			
SIP Parameter	INVITE			
on rarameter	Content-Type: multipart/mixed;boundary=[any boundary name]			
	Someth Type: manaparemines, Seamany -[any Seamany name]			
	[any boundary name]			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	IAM			
	Redirecting number			
	Address presentation restricted indicator			
	presentation restricted			
	Address signal (<i>Diverting user</i>)			
	Original called number Address presentation restricted indicator			
	presentation restricted			
	Address signal			
	Redirection information			
	Original Redirection Reason unknown			
	Redirecting indicator			
	Redirection counter			
	Redirecting reason			
	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			
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A			
	Check: Is a INVITE request received at the interconnection interface?			
	Check: Is an IAM encapsulated in the INVITE?			
	Check: Is the Redirecting number present and the Address presentation			
	restricted indicator is set to 'presentation restricted'?			
	Check: Is the Original called number present and the Address presentation			
	restricted indicator is set to 'presentation restricted'?			
	Check: Is the Redirection number present?			
	Check: Is Redirection information present and the Redirecting reason is set to			
	'Mobile subscriber not reachable'?			
	Repeat this test in reverse direction.			
	•			

7.1.5.6.5 Communication Deflection

Test case number	SS_cd_001			
Test case group	SIP-SIP/Service/CD			
Reference	4.5.2.6/[9]			
SELECTION EXPRESSION	SE 29			
Test purpose	Commu	nication deflection during alertir	ng, basic ru	iles.
	The user	A and user C are in Network A. T	he user B is	in network B and is
	p	with CDa.		
		nat when user A calls user B, the o		
	C. In the	active call state, ensure the prope	erty of speed	h.
Configuration				
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
	_	CDa is performed		
	←	180 Ringing(Call-ID B-A)		
	←	INVITE(Call-ID B-C)	_	
	_	180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)	_	
	-	200 OK INVITE(Call-ID C-B)	→	
	(ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)	_	
		ACK(Call-ID A-B)	→	
		Communication		
0	01	Apply post test routine		
Comments	Check:	CDa is successful.		f an arab
	Check:	In the active call state, ensure th		
	Check:	Is the P-Asserted-Identity preser	it set to the	identity of the originating
	Donoct #	user?		
	Repeat ti	nis test in reverse direction.		

	1			
Test case number		SS_cd_002		
Test case group	SIP-SIP/	Service/CD		
Reference	4.5.2.6/[9	9]		
SELECTION EXPRESSION	SE 29			
Test purpose	Commu	nication deflection immediate, b	asic rules.	
	Ensure tl	nat when user A calls user B which	n deflects th	ne communication towards
	user C in	nmediately (i.e. before alerting star	rts), the cal	I is forwarded 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)	→	
		CDi is performed		
	←	INVITE(Call-ID B-C)		
		180 Ringing(Call-ID C-B)	→	
	←	180 Ringing(Call-ID B-A)		
		200 OK INVITE(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	200 OK INVITE(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Communication		
		Apply post test routine		
Comments	Check:	CDi is successful.		
	Check:	In the active call state, ensure th	e property	of speech.
	Check:	Is the P-Asserted-Identity preser	nt set to the	identity of the originating
		user?		
	Repeat t	Repeat this test in reverse direction.		

Test case number	SS_cd_003		
Test case group	SIP-SIP/Service/CD		
Reference	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 29 AND SE 30		
Test purpose	Communication Deflection immediate response, no notification.		
	The user A and user C are in Network A. The user B is in network B and is provided with CFU, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded message.		
Configuration	Subscription options:		
	Originating user receives notification that his communication has been diverted = No		
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface INVITE(Call-ID A-B) CDi is performed INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) 180 Ringing(Call-ID B-A) Apply post test routine		
Comments	Check: No notification regarding call forwarding in network B is received at the		
	interconnection interface.		
	Check: Is the cause parameter in the last entry is set to '480'.		
	Repeat this test in reverse direction.		

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

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

Test case number	SS cd 006			
Test case group	SIP-SIP/Service/CD			
Reference	4.5.2.6/[9]			
SELECTION EXPRESSION	SE 29 AND SE 30			
Test purpose	Communication Deflection immediate response, diverted-to user does not receive the URI of the served user.			
	The user A and user C are in network A. The user B is in network B and is provided with CFU "Served user allows the presentation of his/her URI to the diverted-to user" = No. Ensure that when user A calls user B which deflects the communication towards user C immediately (i.e. before alerting starts), the call is forwarded to user C, user C is not informed of the forwarding number.			
Configuration	Subscription options:			
_	 Served user allows the presentation of his/her URI to diverted-to user = No 			
SIP Parameter	INVITE Request line contains ';cause=480' History-Info: <sip:userb@networkb?privacy=history&reason=sip;cause=302>;index=1, <sip: userc@networka;cause="480">;index=1.1</sip:></sip:userb@networkb?privacy=history&reason=sip;cause=302>			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CDi is performed INVITE(Call-ID B-C) Apply post test routine			
Comments	Check: A History-Info header is received in the INVITE contains the URI of			
	user B (served user) at the interconnection interface 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.			
		and user C are in network A. The user B is in network B and is		
	1.	th CFU "Served user allows the presentation of his/her URI to		
		user" = Yes.		
		t when user A calls user B which deflects the communication towards		
		rediately (i.e. before alerting starts), the call is forwarded to user C,		
Configuration	user C is informed of the forwarding number. Subscription options:			
Comigaration	 Served user allows the presentation of his/her URI to diverted-to user = Yes 			
SIP Parameter	INVITE			
	Request line contains ';cause=480'			
	History-Info:			
	<pre><sip:userb@networkb?reason=sip;cause=302>;index=1,</sip:userb@networkb?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)		
0		Apply post test routine		
Comments		History-Info header is received in the INVITE contains the URI of		
		ser B (served user) at the interconnection interface. the 'cause' parameter present in the Request line sent to user C		
		liverted-to user) set to '480'?		
		the cause parameter in the last entry is set to '480'?		
		The history entries can be accumulated in "one" History-Info header		
		or each history entry is present in one single History-Info header.		
		test in reverse direction.		

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

Test case number	SS_cd_0	09		
Test case group	SIP-SIP/	Service/CD		
Reference	4.5.2.6/[9	9]		
SELECTION EXPRESSION	SE 29			
Test purpose	Commu	nication Deflection immediate re	esponse, u	ınsuccessful NDUB.
		A and user C are in network A. T		
		nat when user A calls user B, the		cted immediate to user C
	and user	C is network determined user but	sy.	
Configuration				
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
		CDi is performed		
	←	INVITE(Call-ID B-C)		
		486 Busy Here(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	486 Busy Here(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Apply post test routine		
Comments	Check:	The dialogue is terminated by re	eceiving a 4	186 Busy Here.
	Repeat th	nis test in reverse direction		•

Test case number	CC ad 040		
	SS_cd_010		
Test case group	SIP-SIP/Service/CD		
Reference	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 29 AND SE 30 AND [Network A] SE 9		
Test purpose	Communication Deflection immediate response, interaction with a not trusted network.		
	The user A and user C are in network A. The user B is in network B and is provided with CD Originating user receives notification that his communication has been diverted = Yes ("Served user allows the presentation of forwarded to URI to originating user in diversion notification"=Yes, "diverting number is released to the diverted-to user"=Yes. Ensure that when user A calls user B, the call is deflected immediate response to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.		
Configuration			
SIP Parameter	Subscription options: Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to the diverted-to user = No		
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CDi is performed INVITE(Call-ID B-C) 181 Being Forwarded(Call-ID B-A)		
	Apply post test routine		
Comments	Check: No History-Info header is received in the INVITE at the interconnection interface. Check: No History-Info header is received in the 181 Being Forwarded at the		
	interconnection interface.		
	Repeat this test in reverse direction.		

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

Test case number	SS cd 013		
	SIP-SIP/Service/CD		
Test case group Reference			
SELECTION EXPRESSION	[6.5/[24] [Network B] SE 17 AND SE 47 AND SE 55		
Test purpose	SIP-I support. CD performed in Network B, Notification subscription options is set to presentation allowed with redirection number.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part		
	of Network B and is provided with CDi or CDa, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to		
	user number.		
	Ensure that when user A calls user B, the call is deflected to user C, user A is		
	notified of call diversion and informed of the diverted-to number.		
	The notification information is present in the encapsulated ACM contained in the		
	Redirection number and Call diversion information if SIP-I - ISUP/BICC		
	interworking is applicable in Network B.		
Configuration	Subscription options:		
	Calling user receives notification that his call has been diverted (forwarded and file and diverted to the diverted to th		
CID Devementer	or deflected) = yes, with diverted-to user number		
SIP Parameter	183 /180 Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	ACM/CPG		
	Redirection number		
	Address signal (Diverted-to user)		
	Call diversion information		
	Notification subscription options		
	presentation allowed with redirection number		
	Redirecting reason Deflection immediate or Deflection during alerting		
	Generic notification		
	call is diverting		
	[any boundary name]		
Message flow	[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		
	CD is performed		
	← INVITE(Call-ID B-C, IAM)		
	← 183 / 180 (Call-ID B-A, ACM/CPG)		
	Apply post test routine		
Comments	Originating user in Network A establishes a call to user in Network B. Network B		
	performs the diversion to a user in Network A		
	Check: 183 Session Progress is received at the interconnection interface.		
	Check: Is an ACM encapsulated in the 183?		
	Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present?		
	Check: Is 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			
Configuration	ISUP/BICC- SIP-I interworking is applicable in Network A. Subscription options:			
Configuration	 Connected user subscribed to COLR, Permanent = yes 			
SIP Parameter				
SIF Farailleter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name]			
	lany boundary namej Content-Type: application/isup;version=itu-t92			
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required			
	Content-Disposition. signal, nandling=required			
	ANM			
	Redirection number restriction			
	Presentation restricted			
	[any boundary name]			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE(Call-ID A-B), IAM →			
	★ 180 Ringing (Call-ID B-A) in case CDa CD is performed			
	← INVITE(Call-ID B-C)			
	180 Ringing (Call-ID C-B, ACM) →			
	← 180 Ringing (Call-ID B-A)			
	200 OK INVITE (Call-ID C-B, ANM) →			
	ACK (Call-ID B-C)			
	€ 200 OK INVITE (Call-ID B-A)			
	ACK (Call-ID A-B) →			
	Apply post test routine			
Comments	Originating user in Network A establishes a call to user in Network B. Network B			
	performs the diversion to a user in Network A			
	Check: Is a 200 OK INVITE received at the interconnection interface?			
	Check: Is an ANM encapsulated in the 200 OK?			
	Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation			
	restricted'?			
	Repeat this test in reverse direction.			

Test case number	SS_cd_015			
Test case group	SIP-SIP/Service/CD			
Reference	6.7/[24]			
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53			
Test purpose	SIP-I support. CD performed in Network B, No restriction of the Redirection			
rest 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:			
_	 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			
	Redirection number restriction Presentation allowed or Redirection number restriction not present			
	[any boundary name]			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE(Call-ID A-B), IAM → 180 Ringing (Call-ID B-A) in case CDa CD is performed			
•	► INVITE(Call-ID B-C)			
•	180 Ringing (Call-ID C-B, ACM) → 180 Ringing (Call-ID B-A)			
	200 OK INVITE (Call-ID C-B, ANM) → ACK (Call-ID B-C)			
•	← 200 OK INVITE (Call-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	[Network B] SE 17 AND SE 47 AND SE 55		
Test purpose	SIP-I support. CD performed in Network B, Notification of diverted-to user		
1 000 pui pood	Redirecting number 'presentation allowed'.		
	3 p		
	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		
0	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		
CID Dozomator	number information INVITE		
SIP Parameter			
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	IAM		
	Redirecting number		
	Address presentation restricted indicator		
	presentation 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		
	Deflection immediate or Deflection during alerting		
	[any boundary name]		
Message flow	Interconnection Interface SIP (Network B)		
SIP (Network A)			
	INVITE(Call-ID A-B) → 180 Ringing (Call-ID B-A) in case CDa		
	CD 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 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.		
<u> </u>	proposition toot in reverse unconern		

Test case number	SS_cd_017		
Test case group	SIP-SIP/Service/CD		
Reference	7.1/[24]		
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55		
Test purpose	SIP-I support. CD performed in Network B, Notification of diverted-to user		
root pui pood	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 number information		
SIP Parameter	diverting numberinformation 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		
	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		
	Deflection immediate or Deflection during alerting		
	Bollocion inimodiate of Bollocion 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 NVITE(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>vuri of conference focus;method=INVITE > NOTIFY(B1, 100) Content-Type: message/sipfrag SIP/2.0 100 INVITE: Request URI: uri of conference focus From: user B1 NOTIFY(B1, 200) Content-Type: message/sipfrag</uri>
	SIP/2.0 200 OK REFER(user B2) Refer-To: <uri conference="" focus;method="INVITE" of=""> NOTIFY(B2, 100) Content-Type: message/sipfrag SIP/2.0 100 INVITE: Request URI: uri of conference focus From: user B2 NOTIFY(B2, 200) Content-Type: message/sipfrag SIP/2.0 200 OK</uri>

Message flow SIP (Network /	Δ)	Interconnection Interface		SIP (Network B)
		to user B1 from Network A to N	letwork B a	
		to user B2 from Network A to N		
2010011011 0		A establishes a 3PTY conversat		na pat it on noid
	333.	REFER(user B1)	→	
	←	202 Accepted	-	
	←	NOTIFY(B1, 100)		
		200 OK NOTIFY	→	
	←	INVITE(focus, user B1)		
		200 INVITE	→	
	←	ACK		
	←	NOTIFY(B1, 200)		
		200 OK NOTIFY	→	
		BYE(user B1)	→	
	←	200 OK BYE		
		REFER(user B2)	→	
	←	202 Accepted		
	←	NOTIFY(100)		
		200 OK NOTIFY	→	
	←	INVITE(focus, user B2)		
		200 INVITE	→	
	←	ACK		
	←	NOTIFY(B2, 200)		
		200 OK NOTIFY	→	
		BYE(user B2)	→	
	(200 OK BYE		
		Apply post test routine		
Comments	User A e	stablishes a 3PTY conversation a	fter the confi	irmed communication to
	user B1	and B2 are set on hold		
	Check:	The Refer-To header in the REF		
		contains the URI of the conferer	ice focus and	d is the method paramete
		set to 'INVITE'.		
	Check:	The NOTIFY after the REFER re	equest conta	ins the 'SIP/2.0 100'
		message body.		
	Check:	The INVITE request is sent by u		
		focus the Request URI is used f	rom the Refe	er-To header of the
		received REFER request.		IOID/c 6:::
	Check:	The NOTIFY after the REFER re	equest conta	ins the 'SIP/2.0 200 OK'
		message body.		
	Check:	The original session is terminate	ed by user A.	
	Repeat t	his test in reverse direction.		

Test case number	SS_conf_002
Test case group	SIP-SIP/Service/CONF
Reference	4.5.2/[10], 4.7.2.9.7/[20]
SELECTION EXPRESSION	[Network A] SE 12 AND SE 31
Test purpose	3 Party establishment using relNVITE performed by the AS in network A.
	User B1 and user B2 are located in network B, user A is located in network A. A confirmed session from user A to user B1 is set on hold; a confirmed session from user A to user B2 is set on hold. • Ensure that user A can invite user B1 to the conference by sending a reINVITE request.
	 Ensure that user A can invite user B2 to the conference by sending a
	reINVITE request.
Configuration	
SIP Parameter	INVITE <b1> From: <usera> To: <userb1> Call-ID: A-B1 P-Asserted-Identity: <usera></usera></userb1></usera></b1>
	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 NVITE (Call-ID A-B1)
	INVITE(Call-ID A-B2) → 200 INVITE ACK → Apply post test routine
Comments	User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on hold Check: An INVITE is sent to user B1 and user B2 indicating a new IP address in the 'c' line of the SDP. Check: The 'a' line indicates 'sendrecv'.

Test case number	SS conf 003		
	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. 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 catablished'. 		
	 established'; an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established'. 		
Configuration	ISUP/BICC interworking applies in Network A User in Network A is subscribed to the 3PTY supplementary service		
SIP Parameter	INFO <b1></b1>		
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	CPG		
	Generic Notification		
	Conference established		
	INFO <b2></b2>		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	CPG		
	Generic Notification		
	Conference established		
	Interconnection Interface SIP (Network B) ssion from User A in Network A to user B1 in Network B and put it on hold firmed session from User A in Network A to user B2 in Network B INFO(Call-ID A-B1, CPG) 200 INFO		
	INFO(Call-ID A-B2, CPG) → 200 INFO Apply post test routine		
Comments	User A establishes confirmed communication to user B1 in Network B and sets it		
	on hold		
	User A establishes a confirmed communication to user B2 in Network B User A invokes the 3PTY communication		
	Check: Is an INFO request sent to user B1 and user B2 in Network B?		
	Check: Is a ISUP/BICC CPG message encapsulated in the INFO request to both remote users in Network B?		
	Check: Is the Generic Notification parameter in the encapsulated CPG in both INFO set to 'Conference established'?		
	Repeat this test in reverse direction.		

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. 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 Use B1 in Network B and sets it on HOLD. User A establishes a confirmed communication with a User B2 in Network B. User A invokes 3PTY conversation Ensure that when User A disconnects the previous active user: • a BYE request is sent to User B1 in Network B; • an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Conference disconnected'.
Network A SE 17 AND SE 47 AND SE 56
Network A SE 17 AND SE 47 AND SE 56
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 Use B1 in Network B and sets it on HOLD. User A establishes a confirmed communication with a User B2 in Network B. User A invokes 3PTY conversation Ensure that when User A disconnects the previous active user: • a BYE request is sent to User B1 in Network B; • an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Conference'
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 Use B1 in Network B and sets it on HOLD. User A establishes a confirmed communication with a User B2 in Network B. User A invokes 3PTY conversation Ensure that when User A disconnects the previous active user: • a BYE request is sent to User B1 in Network B; • an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Conference
 applies in Network A. User A establishes a confirmed communication with a Use B1 in Network B and sets it on HOLD. User A establishes a confirmed communication with a User B2 in Network B. User A invokes 3PTY conversation Ensure that when User A disconnects the previous active user: a BYE request is sent to User B1 in Network B; an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Conference
B1 in Network B and sets it on HOLD. User A establishes a confirmed communication with a User B2 in Network B. User A invokes 3PTY conversation Ensure that when User A disconnects the previous active user: • a BYE request is sent to User B1 in Network B; • an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Conference
communication with a User B2 in Network B. User A invokes 3PTY conversation Ensure that when User A disconnects the previous active user: • a BYE request is sent to User B1 in Network B; • an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Conference
 Ensure that when User A disconnects the previous active user: a BYE request is sent to User B1 in Network B; an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Conference'
 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'
 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
is encapsulated the Generic Notification is set to 'Conference
· ·
l disconnected
Configuration ISUP/BICC interworking applies in Network A
User in Network A is subscribed to the 3PTY supplementary service SIP Parameter INFO <b2></b2>
Content-Type: application/isup;version=itu-t92
Content-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)
Establish a confirmed session from User A in Network A to user B1 in Network B and put it on hold
Establish a confirmed session from User A in Network A to user B2 in Network B
User A establishes a 3PTY conversation
BYE(Call-ID A-B1, REL) →
← 200 INFO
INFO(Call-ID A-B2, CPG) →
• 200 INFO
Apply post test routine
Comments User A establishes a 3PTY conversation with user B1 and user B2 located in
Network B
User A disconnects the communication with user B1 in Network B (previous on
hold)
Check: Is a BYE request is sent to user B1 in Network B?
Check: Is a ISUP/BICC CPG message encapsulated in the INFO request to
user B2 in Network B?
Check: Is the Generic Notification parameter in the encapsulated CPG in the
INFO sent to user B2 set to 'Conference disconnected'?
Repeat this test in reverse direction.

Test case number Test case group Reference	SS_conf_005		
	SIP-SIP/Service/CONF		
	5.4/[24]		
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56		
	SIP-I/ISUP interworking. Served user splits the 3 Party communication.		
rest purpose	SIF-1/150P interworking. Served user spins the 3 Party communication.		
	Served User A is located in Network A and ISUP/BICC - SIP-I interworking		
	applies in Network A. User A establishes a confirmed communication with a User		
	B1 in Network B and sets it on HOLD. User A establishes a confirmed		
	communication with a User B2 in Network B. User A invokes 3PTY conversation		
	Ensure that when User A splits the 3 PTY communication:		
	an INFO request is sent to User B1 in Network B and a ISUP/BICC CPG		
	is encapsulated the Generic Notification is set to 'Conference		
	disconnected';		
	an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG		
	is encapsulated the Generic Notification is set to 'Conference		
	disconnected'.		
Configuration	ISUP/BICC interworking applies in Network A		
	User in Network A is subscribed to the 3PTY supplementary service		
SIP Parameter	INFO <b1></b1>		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	CPG		
	Generic Notification		
	Conference disconnected		
	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		
	med session from User A in Network A to user B2 in Network B		
	User A establishes a 3PTY conversation		
	INFO(Call-ID A-B1, CPG) →		
	€ 200 INFO		
	200 1111 0		
	INFO(Call-ID A-B2, CPG) →		
	€ 200 INFO		
	Apply post test routine		
Comments	User A establishes confirmed communication to user B1 in Network B and sets it		
Comments			
	on hold		
	User A establishes a confirmed communication to user B2 in Network B		
	Check: Is an INFO request sent to user B1 and user B2 in Network B?		
	Check: Is a ISUP/BICC CPG message encapsulated in the INFO request to		
	both remote users in Network B?		
	Check: Is the Generic Notification parameter in the encapsulated CPG in both		
	INFO set to 'Conference established'?		
	Repeat this test in reverse direction.		

Test case number	SS_conf_006		
Test case group	SIP-SIP/Service/CONF		
Reference	5.4/[24]		
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56		
Test purpose	SIP-I/ISUP interworking. Establishment of aCONF conversation.		
	Served User A is located in Network A and ISUP/BICC - SIP-I interworking applies in Network A. User A establishes a confirmed communication with a User B1 in Network B and invokes the CONF communication.		
	Ensure that when User A invokes the CONF communication:		
	an INFO request is sent to User B1 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established' when the conference is invoked.		
	User A establishes a confirmed communication with a User B2 in Network B. Ensure when User A adds the user B2 to the established conference:		
	 an INFO request is sent to User B1 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'Other party; an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG 		
	is encapsulated the Generic Notification is set to 'conference established' when the user is added to the conference.		
Configuration	ISUP/BICC interworking applies in Network A		
OID Demand	User in Network A is subscribed to the 3PTY supplementary service		
SIP Parameter	INFO1 <b1> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required</b1>		
	CPG		
	Generic Notification		
	conference established		
	INFO2 <b1></b1>		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	CPG		
	Generic Notification		
	Other party added		
	INFO <b2></b2>		
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	CPG		
	Generic Notification		
	conference established		
Message flow SIP (Network A) Establish a confi	Interconnection Interface SIP (Network B) irmed session from User A in Network A to user B1 in Network B		
	User A establishes a CONF conversation		
	INFO1(Call-ID A-B1, CPG) → 200 INFO		
Establish a confirmed so	ession from User A in Network A to user B2 in Network B and add to the conference		
	INFO2(Call-ID A-B2, CPG)		
	INFO(Call-ID A-B2, CPG) → 200 INFO		
	Apply post test routine		

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

Test case number	SS_conf_007		
Test case group	SIP-SIP/Service/CONF		
Reference	5.4/[24]		
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56		
Test purpose	SIP-I/ISUP interworking. Isolation and Reattachment of one party of the conference.		
	Served User A is located in Network A and ISUP/BICC - SIP-I interworking applies in Network A. User A invokes a CONF communication with user B1 and user B2 in Network B.		
	Ensure that when User A isolates one remote party (B1) from the CONF communication:		
	 an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'isolated' in the encapsulated ISUP/BICCCPG; an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party isolated' in the encapsulated ISUP/BICCCPG. 		
	Ensure that when User A reattaches one remote party (B1) to the CONF communication:		
	 an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'reattached' in the encapsulated ISUP/BICCCPG; an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party reattached' in the encapsulated ISUP/BICCCPG. 		
Configuration	ISUP/BICC interworking applies in Network A User in Network A is subscribed to the 3PTY supplementary service		
SIP Parameter	INFO1 <b1> CPG Generic Notification= isolated</b1>		
	INFO2 <b1> CPG Generic Notification= Other party isolated</b1>		
	INFO1 <b2> CPG Generic Notification= reattached</b2>		
	INFO2 <b2> CPG Generic Notification= Other party reattached</b2>		

Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
Establish a	CONF com	munication with User B1 and User	B2 in Network B
	User A isola	tes User B1 from the CONF conver	sation
		INFO1(Call-ID A-B1, CPG)	→
	←	200 INFO	
		<mark>INFO1</mark> (Call-ID A-B2, <mark>CPG</mark>)	→
	←	200 INFO	
•	User A reatta	aches User B1 to the CONF conver	sation
		INFO2(Call-ID A-B2, CPG)	→
	←	200 INFO	
		INFO2(Call-ID A-B2, CPG)	→
	←	200 INFO	
		Apply post test routine	
Comments	User A I	nvokes a CONF conversation with Us	er B1 and User b2 in Network B
	User A s	plits user B1 in Network B from the C	ONF conversation
	Check:	•	
		to 'isolated' in the encapsulated CP0	
	Check:	Is an INFO request sent to user B2	
		to 'Other party isolated' in the encap	
	User A reattaches user B1 in Network B to the CONF conversation		
	Check:		
		to 'reattached' in the encapsulated (
	Check:	Is an INFO request sent to user B2	
		to 'Other party reattached' in the end	capsulated CPG?
	Repeat t	his test in reverse direction	

Tost case number	CC conf 000
Test case number	SS_conf_008
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	SIP-I/ISUP interworking. Splitting and Adding of a party.
	Served User A is located in Network A and ISUP/BICC - SIP-I interworking applies in Network A. User A invokes a CONF communication with user B1 and user B2 in Network B.
	Ensure that when User A split one remote party (B1) from the CONF communication:
	an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'conference disconnected' in the encapsulated ISUP/BICCCPG;
	 an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party split' in the encapsulated ISUP/BICCCPG.
	Ensure that when User A adds one remote party (B1) to the CONF communication:
	 an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'Conference established' in the encapsulated ISUP/BICCCPG;
	 an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party added' in the encapsulated ISUP/BICCCPG.
Configuration	ISUP/BICC interworking applies in Network A User in Network A is subscribed to the 3PTY supplementary service
SIP Parameter	INFO1 <b1></b1>
	CPG Generic Notification= conference disconnected
	INFO2 <b1></b1>
	CPG
	Generic Notification=Other party split
	INFO1 <b2></b2>
	Generic Notification=Conference established
	INFO2 <b2></b2>
Manager Classes	Generic Notification= Other party added
	Interconnection Interface SIP (Network B) CONF communication with User B1 and User B2 in Network B
U	ser A isolates User B1 from the CONF conversation
	INFO1(Call-ID A-B1, CPG) → 200 INFO
	INFO1(Call-ID A-B2, CPG) →
	← 200 INFO
u	ser A reattaches User B1 to the CONF conversation
J	INFO2(Call-ID A-B2, CPG) →
	← 200 INFO INFO2(Call-ID A-B2, CPG) →
	← 200 INFO
Comments	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-c	cb 001		
Test case group		SIP-SIP/Service/ACR-CB		
Reference		4.5.2.6/[12]		
SELECTION EXPRESSION	SE 32	1		
Test purpose	Call Bar	Call Barring performed in network B for user B.		
		s located in network A and user E		in network B and is
		hat a communication from user A		d in network B by sending a
	603 Decl	ine due to the Call Barring service	ce of user E	3.
Configuration	User B is	s subscribed to the incoming Cal	Barring se	ervice (e.g. user A in a black
	list)			
SIP Parameter	INVITE			
		P-Asserted-Identity: <uri of="" th="" u<=""><th>ser A></th><th></th></uri>	ser A>	
Message flow				
SIP (Network A)		Interconnection Interface		SIP (Network B)
,		INVITE	→	,
	←	603 (Decline)		
		ÀCK	→	
Comments	Check:	Is the P-Asserted-Identity pres	ent?	
	Check:	Is the communication rejected	by sending	g a 603 (Decline) final
	response	e sent to user A?	,	. ,
		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 SIP Parameter	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. User B is subscribed to the Anonymous Communication Rejection service INVITE		
	P-Asserted-Identity: <uri a="" of="" user=""> Privacy: id</uri>		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →		
	← 433 (Anonymity Disallowed) ACK →		
Comments	Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the communication rejected by sending a 433 (Anonymity Disallowed) final response sent to user A? Repeat this test in reverse direction.		

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

7.1.5.9 Closed User Group (CUG)

Test case number	SS_cug_001		
Test case group	SIP-SIP/Service/CUG		
Reference	4.5.2.4/[13]		
SELECTION EXPRESSION	SE 34		
Test purpose	Originating user +OA to terminating user no CUG.		
	An originating user in a CUG Outgoing Access allowed calls to a user not in a		
	CUG. The session establishment is successful.		
Configuration	Originating user: CUG, outgoing access allowed		
SIP Parameter	INVITE:		
	Content-Type: application/vnd.etsi.cug+xml		
	<cug></cug>		
	<networkindicator>01<!-- networkIndicator</th--></networkindicator>		
	<networkindicator>23<!-- networkIndicator</th--><th>nom (Codo)</th></networkindicator>	nom (Codo)	
	<cuginterlockbinarycode>0F03</cuginterlockbinarycode> 0F03		
	<cugcommunicationindicator>10</cugcommunicationindicator>		
Message flow	<:cug>		
SIP (Network A)	Interconnection Interface SII	P (Network B)	
	INVITE →	(1011101112)	
	← 180 Ringing		
	Apply post test routine		
Comments	Check: Is the Content-Type in The INVITE set to		
	application/vnd.etsi.cug+xml?		
	Check: Contains the XML body in the INVITE a 'cug' element?		
	Check: Contains the XML body in the INVITE a 'networkIn	dicator' element as	
	a 'cug' child element?		
	Check: Contains the XML body in the INVITE a 'cugInterlockBinaryCode'		
	element as a 'cug' child element?		
	Check: Contains the XML body in the INVITE a 'cugCommunicationIndicator'		
	element set to '10' as a 'cug' child element?		
	Check: Is the session setup not rejected?		
	Repeat this test in reverse direction. NOTE: The networkIndicator element value and the cugln	otorlook Dinory Codo	
		nenockomarycode	
	element value are examples.		

Test case number	SS_cug_	002	
Test case group	SIP-SIP/Service/CUG		
Reference	4.5.2.4, 4.5.2.10/[13]		
SELECTION EXPRESSION	SE 34		
Test purpose	Originating user -OA to terminating user no CUG.		
l'est pui pose	Originat	Originating user -OA to terminating user no COG.	
	An origin	ating user in a CUG Outgoing Access not allowed calls to a user not in	
	a CUG. The session establishment is not successful, a 403 (Forbidden)		
	response is sent.		
Configuration	Originating user: CUG, outgoing access not allowed		
SIP Parameter	INVITE:		
	Content-Type: application/vnd.etsi.cug+xml		
		ent-Disposition:;handling= required	
	00111	Six Disposition:, namating—required	
	<pre></pre> <pre><cug> </cug></pre> <pre><networkindicator>01</networkindicator></pre> /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	
		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'	
	0	element as a 'cug' child element?	
	Check:	Contains the XML body in the INVITE a 'cugCommunicationIndicator'	
	Chaola	element set to '11' as a 'cug' child element?	
	Check:	Is the session setup rejected? A 403 (Forbidden) final response is sent	
by the terminating network?		by the terminating network? his test in reverse direction.	
		nis iest in reverse nireminn	
	NOTE:	The networkIndicator element value and the cugInterlockBinaryCode element value are examples.	

Test case number	SS_cug_003		
Test case group	SIP-SIP/Service/CUG		
Reference	4.5.2.4, 4.5.2.10/[13]		
SELECTION EXPRESSION	SE 34		
Test purpose	Originating user -OA to terminating user -IA.		
		ating user in a CUG Outgoing Access n	
	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		
		etwork A and user in network B are in the	ne same CUG
SIP Parameter	INVITE:		
		ent-Type: application/vnd.etsi.cug+xml	
	Conte	ent-Disposition:;handling= required	
	<cl< th=""><th>IQ></th><th></th></cl<>	IQ>	
		networkIndicator>01 <th>ator</th>	ator
		networkIndicator>23 <th></th>	
		cugInterlockBinaryCode>0F03 <th></th>	
		cugCommunicationIndicator>11 <th></th>	
	<cugoommunicationinalcator> 1 </cugoommunicationinalcator> <cug></cug>		
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
	_	INVITE	•
	←	180 Ringing	
	Ta	Apply post test routine	
Comments	Check:	Is the Content-Type in The INVITE set application/vnd.etsi.cug+xml?	: to
	Check:	Is the handling parameter in the Conte	ent-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 '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	•
		element set to '11' as a 'cug' child elen	nent?
	Check:	Is the session setup not rejected?	
		his test in reverse direction.	
	NOTE:	The networkIndicator element value a	nd the cugInterlockBinaryCode
		element value are examples.	

Test case number	SS_cug_	004	
Test case group	SIP-SIP/Service/CUG		
Reference	4.5.2.4, 4.5.2.10/[13]		
SELECTION EXPRESSION	SE 34		
Test purpose	Originat	ing user in a CUG to terminating user -IA.	
	An originating user in a CUG calls to a user in a different CUG Incoming Access not allowed. The session establishment is not successful, a 403 (Forbidden) response is sent.		
Configuration	User in network A and user in network B are not in the same CUG		
	Termina	ting user: CUG incoming access not allowed	
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml Content-Disposition:;handling= requiredv		
	<pre> <cug> <networkindicator>01</networkindicator>230F03 <cugcommunicationindicator></cugcommunicationindicator> <cug></cug></cug></pre>		
Message flow			
SIP (Network A)		Interconnection Interface SIP (Network B)	
		INVITE ->	
	←	403 (Forbidden)	
_	T	ACK →	
Comments	Check: Check: 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? Contains the XML body in the INVITE a 'networkIndicator' element as	
	Check:	a 'cug' child element? Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element?	
	Check:	Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '10' or '11'as a 'cug' child element?	
	Check:	Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network?	
	Repeat this test in reverse direction.		
	NOTE:	The networkIndicator element value and the cugInterlockBinaryCode	
		element value are examples.	

Test case number	SS_cug_	005		
Test case group	SIP-SIP/Service/CUG			
Reference	4.5.2.10/[13]			
SELECTION EXPRESSION	SE 34			
Test purpose	Originati	ng user no CUG to terminatin	g user +IA.	
		ating user not in a CUG calls to The session establishment is su		CUG Incoming Access
Configuration	Terminating user: CUG incoming access allowed			
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE	→	
	←	180 Ringing		
		Apply post test routine		
Comments	Check:	Is the session setup rejected?	A 403 (Forbi	dden) final response is sent
		by the terminating network.		
	Repeat th	nis 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 Inconallowed. The session establishment is not successful, a 403 (Foresponse is sent.		
Configuration	User in Network B in a CUG incoming access not allowed		
SIP Parameter			
Message flow SIP (Network A)	INVITE → 403 (Forbidden)	(Network B)	
Comments	ACK → Check: Is the session setup rejected? A 403 (Forbidden) final by the terminating network. Repeat this test in reverse direction.	al response is sent	

Test case number	SS_cug_	007	
Test case group	SIP-SIP/Service/CUG		
Reference	4.5.2.4/[13]		
SELECTION EXPRESSION	SE 34	oj.	
Test purpose		ng user -OA, network B does not support CUG.	
l est purpose	Originating 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:		
	Content-Type: application/vnd.etsi.cug+xml		
	Conte	ent-Disposition:;handling= required	
	<cl< th=""><th></th></cl<>		
		networkIndicator>01	
	<networkindicator>23</networkindicator>		
		cugInterlockBinaryCode>0F03	
	<cugcommunicationindicator>10</cugcommunicationindicator>		
Message flow	\ 00	97	
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	
	01100111	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'	
	O 1 1	element set to '11' as a 'cug' child element?	
	Check:	Is the session setup rejected by sending an unsuccessful final	
	Donast "	response?	
	NOTE:	nis test in reverse direction. The networkIndicator element value and the cugInterlockBinaryCode	
	NOTE:	·	
		element value are examples.	

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

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

Test case number	SS_cug_010	
Test case group	SIP-SIP/Service/CUG	
Reference	7.1/[24]	
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 58) AND ([Network B] SE 17 AND SE 47 AND SE 58)	
Test purpose	SIP-I/ISUP interworking. CUG call with outgoing access not allowed (both user in the same CUG).	
	User A in a CUG is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. User B is located in the PSTN/PLMN part and SIP-I - ISUP/BICC interworking applies in the same CUG. Ensure that when user A is in a CUG 'outgoing access not allowed' calls user B in Network B. The call is successful. There is a Optional forward call indicator the CUG Call Indicator Outgoing access not allowed present in the encapsulated IAM sent to Network B.	
Configuration	User in PSTN/PLMN part of Network A in a CUG outgoing access not allowed User in PSTN/PLMN part of Network B in a CUG	
SIP Parameter	User A and User B are in the same CUG INVITE	
Message flow	Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Optional Forward call indicator CUG Call Indicator Outgoing access not allowed CUG interlock code[any boundary name]	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 180 Ringing	
Comments	User A in the PSTN part of Network A calls user B in the PST/PLMN part of Network B Check: Is an IAM encapsulated in the INVITE request sent from Network A to Network B? Check: Is the Optional forward call indicator present, the CUG Call Indicator is set to 'Outgoing access not allowed'? Check: Is the CUG interlock code parameter present in the encapsulated IAM? Check: Is the call setup successful? Repeat this test in reverse direction.	

Test case number	SS_cug_011	
Test case group	SIP-SIP/Service/CUG	
Reference	7.1/[24]	
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 58) AND ([Network B] SE 17 AND SE	
	47 AND SE 58)	
Test purpose	SIP-I/ISUP interworking. CUG call to a CUG user incoming access not	
	allowed (both user in the same CUG).	
	User A in a CUG is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. User B is located in the PSTN/PLMN part and SIP-I - ISUP/BICC interworking applies in the same CUG. Ensure that when user A is in a CUG 'outgoing access not allowed' calls CUG user B in Network B. The call is successful. There is a Optional forward call indicator the CUG Call Indicator Outgoing access not allowed present in the encapsulated IAM sent to Network B.	
Configuration	User in PSTN/PLMN part of Network A in a CUG outgoing access not	
	 allowed User in PSTN/PLMN part of Network B in a CUG incoming access not allowed 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	
Comments	User A in the PSTN/PLMN part of Network A calls user B in Network B User B in the PSTN/PLMN part of Network B. Check: Is an IAM encapsulated in the INVITE request sent from Network A to Network B?	
	Check: Is the Optional forward call indicator present, the CUG Call Indicator is	
	set to 'Outgoing access not allowed'? Check: Is the CUG interlock code parameter present in the encapsulated IAM?	
	Check: Is the call setup successful?	
	Repeat this test in reverse direction.	

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

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

Test case number	SS_cug_014		
Test case group	SIP-SIP/Service/CUG		
Reference	7.1/[24]		
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 58		
Test purpose	SIP-I/ISUP interworking. Call to a CUG user incor	ning access allowed.	
	User A is located in Network A. User B is located in t SIP-I - ISUP/BICC interworking applied. Ensure that B Incoming access allowed in Network B. The call is	when user A calls CUG user	
Configuration	 User in PSTN/PLMN part of Network B in a CUC 	G incoming access allowed	
SIP Parameter		•	
Message flow SIP (Network A)	Interconnection Interface INVITE →	SIP (Network B)	
0	← 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.		

Test case number	SS 044 (202		_
	SS_cw_002 SIP-SIP/Service/CW			
Test case group Reference				
	4.5.5.2/[15] SE 35 AND SE 36			
SELECTION EXPRESSION				
Test purpose	Call rejected after timeout TAS-CW.			
	the comm Ensure the Waiting in or UDUB	s located in network A, user B is lo munication Waiting service. hat when user A calls user B, user ndication' in the 180 Ringing provi b. After timeout TAS-CW network E ble) response toward user A and t	A receives sional resposed sends a 4	the 'communication onse if the user B is NDUB 80 (Temporarily
Configuration				
SIP Parameter	180: Alert-Info: <urn:alert:service:call-waiting> 480: Reason: Q.850 :cause=19</urn:alert:service:call-waiting>			
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		
		value set to ' <urn:alert:service:ca< th=""><th></th><th></th></urn:alert:service:ca<>		
	Check: Is a Reason header present in the 480 Response and is the protocol is			
		set to 'Q.850' and the cause para	ameter set	to '19'?
	Repeat th	his test in reverse direction.		

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.	
	Llean A is leasted in patriant A was D is leasted in the DCTN/DLMN part of	
	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	
SIF Farailleter	Content-Type: application/isup;version=itu-t92	
	Content-Type: application/isup,version=itu-is2	
	Content-Disposition: signal, nanding=required	
	ACM	
	Backward call indicator	
	Called party's status indicator	
	subscriber free	
	Generic notification	
	Generic notification Notification indicator	
Message flow	Notification indicator call is a waiting call	
Message flow SIP (Network A)	Notification indicator call is a waiting call Interconnection Interface SIP (Network B)	
	Notification indicator call is a waiting call Interconnection Interface SIP (Network B) INVITE →	
	Notification indicator call is a waiting call Interconnection Interface SIP (Network B) INVITE → 180 Ringing	
SIP (Network A)	Notification indicator call is a waiting call Interconnection Interface SIP (Network B) INVITE → 180 Ringing Apply post test routine	
	Notification indicator call is a waiting call Interconnection Interface SIP (Network B) INVITE → 180 Ringing Apply post test routine Check: Is an ISUP/BICC ACM present in the 180 provisional response and	
SIP (Network A)	Notification indicator call is a waiting call Interconnection Interface SIP (Network B) INVITE → 180 Ringing Apply post test routine	

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.	
l set pui pees	on reappoint can training management to their encapositated of or	
	User A is located in network A, user B is located in the PSTN/PLMN part of	
	network B and subscribed to the Call Waiting service.	
	Ensure that when user A calls user B, an encapsulated ISUP/BICC CPG Generic	
	notification 'call is a waiting call' is present in the 180 Ringing provisional	
	response if the user B is NDUB.	
Configuration	User B subscribed to the CW service	
SIP Parameter	180	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	CPG	
	Event information	
	Event indicator	
	ALERTING	
	Generic notification	
	Notification indicator	
Message flow	call is a waiting call	
SIP (Network A)	Interconnection Interface SIP (Network B)	
On (Network A)	INVITE -	
	← 183 Session Progress (ACM)	
	← 180 Ringing (CPG)	
	Apply post test routine	
Comments	Check: Is an ISUP/BICC CPG present in the 180 provisional response and the	
	Generic notification is set to 'call is a waiting cal'?	
	Repeat this test in reverse direction.	

7.1.5.11 Explicit Communication Transfer (ECT)

Test case number	SS_ect_001	
Test case group	SIP-SIP/Service/ECT	
Reference	4.5.2/[11]	
SELECTION EXPRESSION	[Network A] SE 37 AND [Network A] SE 11 AND [Network A] SE 49	
Test purpose	Blind/assured transfer using the REFER method.	
	-	
	User A is located in network A, user B and user C are located in network B. User	
	A invokes ECT to transfer a session with user B to user C.	
	 Ensure that a REFER request is sent from network A to network B in the 	
	dialogue with user B. The 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	DEFED: Descript LIDI address of comp	
SIP Parameter	REFER: Request URI address of user B Refer-To: <uri ect-as="" of="">; method=invite</uri>	
	Refer-10. <ort ec1-a5="" of="">, method=invite</ort>	
	INVITE1 Request URI address of ECT-AS	
	Troquest of traduces 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	r 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 0112	
	← INVITE1 (ECT-AS)	
	INVITE2 (user C)	
	€ 200 OK ÎNVITE	
	ACK →	
	200 OK INVITE →	
	← ACK	
	← NOTIFY (200)	
	200 OK NOTIFY →	
CASE Assured transfer		
CASE		
	BYE (A-B) → 200 OK BYE	
	Apply post test routine	
	Apply post tost routine	

Comments	Check:	Is a REFER request is sent network B, the Refer-To header is set to
		the URI of the ECT-AS in network A and a method parameter is present set to 'INVITE'?
	Check:	Is a NOTIFY request sent to network A containing sipfrag body set to
		'SIP/2.0 100 Trying' and if Blind transfer is applicable the session from user A to user B is terminated by user A?
	Check:	Is an INVITE request sent to network A the Request line is set to the address of the ECT-AS in network A?
	Check:	Is an INVITE request is sent to network B the Request is set to the address of user C?
	Check:	When the session from user B to user C is confirmed a NOTIFY request is sent to network A containing sipfrag body set to 'SIP/2.0 200 OK' and if Assured transfer is applicable the session from user A to user B is terminated by user A?
		Ensure the property of speech between user B and user C.
	Repeat t	his test in reverse direction.

Test case number	SS_ect_0	nn2	
Test case group		Service/ECT	
Reference			
	4.5.2/[11]		11 AND INstructor ALSE FO
Test purpose Configuration SIP Parameter	User A is A invokes It is a series of the s	SECT to transfer a session with Ensure that a REFER request is dialogue with user B. The URI in address of the ECT AS in netwon INVITE'. Ensure that an INVITE request he Request URI is set to the act from the set to the act he Request URI is set to the act the Request URI is set to the act the Request URI is set to the act the set to	R method. Ind user C are located in network B. User in user B to user C. It is sent from network A to network B in the in the Refer-To header is set to the pork A and the method parameter is set to the is sent from network B to network A and didress of the ECT AS in network A. It is sent from network A to network B and didress of user C and a Replaces header in identifiers of the session A - C.
Message flow	Requ	Request URI address of user (ire: replaces aces: <session a-c=""></session>	C
A confir	med sess	ion is established between usion is established between uses ECT to transfer the session REFER 202 Accepted	ser A and user C
	←	NOTIFY (100) 200 OK NOTIFY	→
	←	INVITE1 (ECT-AS) INVITE2 (user C) 200 OK INVITE ACK	→
	-	200 OK INVITE ACK NOTIFY (200) 200 OK NOTIFY	→
	←	BYE (A-B) 200 OK BYE)
	←	BYE (A-C) 200 OK BYE Apply post test routine	→
Comments	Check:		etwork B, the Refer-To header is set to
	Check:	the URI of the ECT-AS in network present set to 'INVITE'? Is an INVITE request sent to raddress of the ECT-AS in network an INVITE request is sent to address of user C and a Replacement.	network A and a method parameter is network A the Request line is set to the work A? To network B the Request is set to the aces header is present contains the
	Check: Check: Repeat th	session identifiers of the sessi Is the session A - B and the se Ensure the property of speech is test in reverse direction.	ession A - C terminated?

Test case number	SS ect (003		
Test case group		Service/ECT		
Reference], 4.7.2.9.7/[20]		
SELECTION EXPRESSION		A] SE37 AND NOT [Network A]	SE 12 AND [Network A] SE 49	
Test purpose	Blind/ass	Blind/assured transfer using the 3pcc method.		
	invokes E • E	ECT to transfer a session with us Ensure that the network A establi	ishes a session to user C. a reINVITE to update the session	
Configuration				
SIP Parameter	INVITE1	Request URI address of user C		
	SDP c= m	Request URI address of user B IN IP4/6 [new IP address] audio [new port] RTP/AVP [new new attributes]		
Message flow		-		
SIP (Network A)	_	Interconnection Interface	SIP (Network B)	
		ion is established between use		
l	ser A Invok	tes ECT to transfer the session		
	←	INVITE1 (user C)	→	
	-	180 Ringing 200 OK INVITE		
	•	ACK	→	
		7.6.1	-	
	←	INVITE2 (user B) 200 OK INVITE	→	
		ACK	→	
Comments	Check:	Apply post test routine	network A to user C to establish a	
	CHECK.	dialogue between network A an		
	Check:		ork A to user B update the session	
	J.10011.	parameter in the SDP?	on, the door b apadio the cooler	
	Repeat th	nis test in reverse direction.		
	1			

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.
	Hear A is legated in naturals A year B and year C are legated in naturals B
	User A is located in network A, user B and user C are located in network B User A invokes ECT to transfer a session with user B to user C.
	Ensure that the network A sends a reINVITE to update the session The second of the second of the session of the second
	between user A and user B (SDP: IP address, port and codec).
	Ensure that the network A sends a reINVITE to update the session
	between user A and user C (SDP: IP address, port and codec).
Configuration	
SIP Parameter	INVITE1: Request URI address of user C
	SDP
	c=IN IP4/6 [new IP address]
	m=audio [new port] RTP/AVP [new codec list]
	a=[new attributes]
	THE PARTY OF THE P
	INVITE2: Request URI address of user B
	SDP IN ID 10 10 10 10 10 10 10 10 10 10 10 10 10
	c=IN IP4/6 [new IP address]
	m=audio [new port] RTP/AVP [new codec list]
	a=[new attributes]
Message flow	Dip (Not on 1 p)
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
USC	er A invokes ECT to transfer the session to user C
	INVITE1 (user B) →
	← 200 OK INVITE
	ACK →
	INIVITED (year C)
	INVITE2 (user C) → 200 OK INVITE
	-
Comments	Apply post test routine Check: Is a reINVITE is sent from network A to user B update the session
Comments	· ·
	parameter in the SDP. Check: Is a reINVITE is sent from network A to user C update the session
	parameter in the SDP.
	Repeat this test in reverse direction.

Test case number	SS_ect_0	005
Test case group		Service/ECT
Reference	5.4.3.2/[2	24]
SELECTION EXPRESSION		A] SE 17 AND SE 47 AND SE 60
Test purpose		port. Call Transfer invoked in active state, call was previous on
cot purpose	HOLD.	opera. Can mander invented in active state, can was provided on
		JP - SIP-I interworking applies in the originating network User A and C ed in network A and user B is located in network B.
	Ensure th	nat an User A can successfully invoke the ECT supplementary service
		sfer the call with User B to User C in active state.
Configuration		subscribed to the Explicit Call Transfer supplementary service
SIP Parameter	INVITE	
		ontent-Type: multipart/mixed;boundary=[any boundary name]
		[any boundary name]
		ontent-Type: application/sdp
	a=	=sendrecv
		[any boundary name]
		ontent-Type: application/isup;version=itu-t92
		ontent-Disposition: signal;handling=required
	F/	AC
		Generic Notification
		Call transfer active
		Call transfer number
		[any boundary name]
Message flow		
SIP (Network A)		Interconnection Interface SIP (Network B)
		stablished between user A and user B and set on hold
U	ser A invok	tes ECT to transfer the session to user C
		INFO (LOP request)
	←	200 OK INFO
	←	INFO (LOP response)
		200 OK INFO →
CASE A		
	_	INVITE (sendrecv; FAC) →
	←	200 OK INVITE
2405 5		ACK →
CASE B		INFO (FAO)
	-	INFO (FAC) →
	←	200 OK INFO
	-	INVITE (sendrecv)
	←	200 OK INVITE
		ACK →
2		Apply post test routine
Comments		n from User A to User B is already established
		ets the User B on hold
		vokes the ECT service
	Check:	Is (optional) an INFO request is sent from Network A to Network B
	1	and an ISUP LOP message is present the Loop prevention indicator
	Chask	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
	Chask	set to 'response'?
	Check:	Is (CASE A) an INVITE request sent and an ISUP FAC message is
	1	present containing a Generic notification indicator is set to 'Call
		transfer active' and in addition the media stream is set to 'sendrecy'?
		CODULOCAL.
	CI 1	
	Check:	Is (CASE B) an INFO request sent and an ISUP FAC message is
	Check:	Is (CASE B) an INFO request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call
	Check:	Is (CASE B) an INFO request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer active'? In addition is an INVITE request sent and the media
		Is (CASE B) an INFO request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer active'? In addition is an INVITE request sent and the media stream is set to 'sendrecv' to resume the held session?
	Check:	Is (CASE B) an INFO request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer active'? In addition is an INVITE request sent and the media stream is set to 'sendrecv' to resume the held session? The content of the FAC in the INVITE request is Equal to the content
	NOTE:	Is (CASE B) an INFO request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer active'? In addition is an INVITE request sent and the media stream is set to 'sendrecv' to resume the held session?

Test case number	SS_ect_0	006	
Test case group		Service/ECT	
Reference	5.4.3.2/[2	24]	
SELECTION EXPRESSION		A] SE 17 AND SE 47 AND SE	60
Test purpose			n alerting state, call was previous on
	HOLD.		3,
	BICC/ISI	JP - SIP-I interworking applies in	n the originating network User A and C
	are locate	ed in network A and user B is lo	cated in network B.
			invoke the ECT supplementary service
	and trans	sfer the call with User B to User	C in alerting state.
Configuration		subscribed to the Explicit Call	
SIP Parameter	INVITE	·	
	C	ontent-Type: multipart/mixed;bo	oundary=[any boundary name]
	!	[any boundary name]	
		ontent-Type: application/sdp	
	a=	=sendrecv	
	!	[any boundary name]	
		ontent-Type: application/isup;ve	ersion=itu-t92
		ontent-Disposition: signal;handl	
	F	AC	
		Generic Notification	
		Call transfer alerting	
		Call transfer number	
		[any boundary name]	
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
		stablished between user A and	
Us	er A invok	ces ECT to transfer the session	n to user C
		INFO (LOP request)	→
	←	200 OK INFO	
	←	INFO (LOP response)	
		200 OK INFO	→
CASE A			
		INVITE (sendrecv; FAC)	→
	←	200 OK INVITE	
		ACK	→
CASE B			
		INFO (FAC)	→
	←	200 OK INFO	
		INVITE (sendrecv)	→
	←	200 OK INVITE	
		ACK	→
		Apply post test routine	
Comments		n from User A to User B is alrea	dy established
		ets the User B on hold	
		n from User A to User C is alrea	dy established
		vokes the ECT service	
	Check:		is sent from Network A to Network B
			present the Loop prevention indicator
		set to 'request'?	
	Check:		is sent from Network A to Network B
			present the Loop prevention indicator
		set to 'response'?	
	Check:		st sent and an ISUP FAC message is
			notification indicator is set to 'Call
			n the media stream is set to 'sendrecv'?
	Check:		sent and an ISUP FAC message is
			notification indicator is set to 'Call
			s an INVITE request sent and the media
		stream is set to 'sendrecv' to re	esume the held session?
	NOTE:	The content of the FAC in the	INVITE request is Equal to the content
		of the FAC in the INFO reques	• •
	Repeat t	his test in reverse direction.	
-			

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		
	SIP-I support. Call Transfer invoked in active state.		
Test purpose	SIP-I support. Call Transfer invoked in active state.		
	DICC/ICLID CID Lintercration and line in the existing time and the existing the existing time.		
	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		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	FAC		
	Generic Notification		
	Call transfer active		
	Call transfer number		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
A confir	med session is established between user A and user C		
Use	er A invokes ECT to transfer the session to user C		
	INFO (LOP request)		
	← 200 OK INFO		
	← INFO (LOP response)		
200 OK INFO →			
	INFO (FAC) →		
	€ 200 OK INFO		
	Apply post test routine		
Comments	A session from User A to User B is already established		
	User A sets the User B on hold		
	A session from User A to User C is already established		
	User A invokes the ECT service		
	Check: Is (optional) an INFO request is sent from Network A to Network B		
	and an ISUP LOP message is present the Loop prevention indicator		
	set to 'request'?		
	Check: Is (optional) an INFO request is sent from Network A to Network B		
	and an ISUP LOP message is present the Loop prevention indicator		
	set to 'response'?		
	Check: Is (CASE B) an INFO request sent and an ISUP FAC message is		
	present containing a Generic notification indicator is set to 'Call		
	transfer active'?		
	NOTE: The content of the FAC in the INVITE request is Equal to the content		
	of the FAC in the INFO request.		
	Repeat this test in reverse direction.		

Test case number	SS_ect_008		
Test case group	SIP-SIP/Service/ECT		
Reference	5.4.3.2/[24]		
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 60		
Test purpose	SIP-I support. Call Transfer invoked in alerting state.		
	DIGO (IGUID GIPLI)		
	BICC/ISUP - SIP-I interworking applies in the originating network User A and B		
	are located in network A and user C is located in network B.		
	Ensure that an User A can successfully invoke the ECT supplementary service		
	and transfer the call with User B to User C in alerting state.		
Configuration	User A is subscribed to the Explicit Call Transfer supplementary service		
SIP Parameter	INFO		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	CPG		
	Generic Notification		
	Call transfer alerting		
	Call transfer number		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	the early dialogue is established between user A and user C		
	er A invokes ECT to transfer the session to user C		
	INFO (LOP request) →		
	€ 200 OK INFO		
	← INFO (LOP response)		
	200 OK INFO →		
	INFO (CPG) →		
	← 200 OK 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.		
	Inopodi uno tost in reverse direction.		

7.1.5.12 Malicious Communication Identification (MCID)

Test case number	SS_mcid_001
Test case group	SIP-SIP/Service/MCID
Reference	4.5.2.5/[18]
SELECTION EXPRESSION	SE 38
Test purpose	Network B sends a MCID request, no response.
Configuration	User A is located in network A, user B is located in network B and subscribed to the Malicious Communication Identification service. When user A call user B and no originating identification is present in the INVITE request, the network B sends an INFO request to network A requesting the originating identity. After timeout of timer TO-ID the network B sends the 180 Ringing response. User B is subscribed to the MCID service
SIP Parameter	INFO:
	<:mcid> <:request> <:McidRequestIndicator>01 :McidRequestIndicator <:HoldingIndicator > :HoldingIndicator :request :mcid
Message flow SIP (Network A)	Interconnection Interface INVITE INFO 200 OK INFO Timeout T _{O-ID} 180 Ringing Apply post test routine
Comments	Check: Is an INFO request sent to network A? Check: Is the McidRequestIndicator element set to ,01'? Check: is a 200 OK INFO response sent to network B? Repeat this test in reverse direction.

Test case number	SS_mcid	002		
Test case group		Service/MCID		
Reference	4.5.2.5/[1			
SELECTION EXPRESSION		ND SE 47		
Test purpose		B sends a MCID request, MCID	response.	
Tool purpose	THOU THE	D condo a moio request, moio	response.	
	PSTN us	er A is located in network A, user	B is located in network B and	
		ed to the Malicious Communication		
	When us	er A call user B and no originating	identification is present in the INVITE	
		the network B sends an INFO req		
	originatin	ig identity. After receipt of an INFO	request from network A the network	
	B sends	the 180 Ringing response.		
Configuration	User B s	ubscribed to the MCID service		
-	User A is	a ISDN or POTS user in the PST	N of network A	
SIP Parameter	INFO:			
		ncid>		
		.:request>		
		<:McidRequestIndicator>01 </th <th></th>		
		<:HoldingIndicator > :Hold</th <th>ingIndicator></th>	ingIndicator>	
		:request>		
	:r</th <th>mcid></th> <th></th>	mcid>		
	INIEO			
	INFO:	a a i al		
	<:mcid>			
	<:response>			
	<:McidResponseIndicator>01 :McidResponseIndicator <:HoldingProvidedIndicator> :HoldingProvidedIndicator			
	<:OrigPartyIdentity>any URI :OrigPartyIdentity			
	<:OrigPartyIdentitySarry OrXIZ/OrigPartyIdentityS			
		true/false	10112	
	1 .	:OrigPartyPresentationRestric</th <th>tion></th>	tion>	
		:response>		
		mcid>		
Message flow	•			
SIP (Network A)		Interconnection Interface	SIP (Network B)	
		INVITE	→	
	←	INFO		
		200 <mark>OK I</mark> NFO	→	
		INFO	→	
	(200 OK INFO		
	<	180 Ringing		
	+	180 Ringing Apply post test routine		
Comments	← Check:	180 Ringing Apply post test routine Is an INFO request sent to netw		
Comments	Check:	180 Ringing Apply post test routine Is an INFO request sent to netw Is the McidRequestIndicator ele	ment set to ,01'?	
Comments	Check: Check: Check:	180 Ringing Apply post test routine Is an INFO request sent to netw Is the McidRequestIndicator ele Is a 200 OK INFO response sen	ment set to ,01'? t to network B?	
Comments	Check: Check: Check: Check:	180 Ringing Apply post test routine Is an INFO request sent to netw Is the McidRequestIndicator ele Is a 200 OK INFO response sen Is an INFO request sent to netw	ment set to ,01'? t to network B? ork B?	
Comments	Check: Check: Check: Check: Check:	180 Ringing Apply post test routine Is an INFO request sent to netw Is the McidRequestIndicator ele Is a 200 OK INFO response sen Is an INFO request sent to netw Is the McidResponseIndicator e	ment set to ,01'? t to network B? ork B? ement set to ,01'?	
Comments	Check: Check: Check: Check: Check: Check:	Apply post test routine Is an INFO request sent to netw Is the McidRequestIndicator ele Is a 200 OK INFO response sen Is an INFO request sent to netw Is the McidResponseIndicator e Is the OrigPartyIdentity element	ment set to ,01'? t to network B? ork B? ement set to ,01'? present in the response element?	
Comments	Check: Check: Check: Check: Check: Check: Check:	Apply post test routine Is an INFO request sent to netw Is the McidRequestIndicator ele Is a 200 OK INFO response sen Is an INFO request sent to netw Is the McidResponseIndicator e Is the OrigPartyIdentity element Is a 200 OK INFO response sen	ment set to ,01'? t to network B? ork B? ement set to ,01'? present in the response element? t to network A?	
Comments	Check: Check: Check: Check: Check: Check: Check: A INFO r	Apply post test routine Is an INFO request sent to netw Is the McidRequestIndicator ele Is a 200 OK INFO response sen Is an INFO request sent to netw Is the McidResponseIndicator e Is the OrigPartyIdentity element Is a 200 OK INFO response sen request containing a mcid response	ment set to ,01'? t to network B? ork B? ement set to ,01'? present in the response element?	
Comments	Check: Check: Check: Check: Check: Check: Check: A INFO r	Apply post test routine Is an INFO request sent to netw Is the McidRequestIndicator ele Is a 200 OK INFO response sen Is an INFO request sent to netw Is the McidResponseIndicator e Is the OrigPartyIdentity element Is a 200 OK INFO response sen	ment set to ,01'? t to network B? ork B? ement set to ,01'? present in the response element? t to network A?	

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

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

7.1.5.13 Message Waiting Indication (MWI)

Test case number	SS_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	Initia	Initial subscription of a Voicemail box.		
		The Voicemail owner is in network A, his Voicemail box is located in network B.		
		re that a Voicemail owner is able to a	activate his	Voicemail box.
Configuration		Voicemail in network B		
		email owner in network A		
SIP Parameter	SUB	CRIBE		
		Event: message-summary		
		Expires: [any value]		
		Accept: application/simple-mes	ssage-sumn	nary
	NOT	NOTIFY		
	INOT	Subscription-State: active;expi	roe-lany va	luol
		Event: message-summary	ies=įaity va	iuej
Message flow		Event. message-summary		
SIP (Network A)		Interconnection Interface		SIP (Network B)
((((((((((((((((((((SUBCRIBE	→	· (· · · · · · · · · · · · · · · · · ·
	←	200 OK SUBSCRIBE		
	←	NOTIFY		
	_	200 OK NOTIFY	→	
	←	200 OK BYE		
	,	NOTIFY		
	←	NOTIFY 200 OK NOTIFY	→	
		Apply post test routine	7	
Comments	Check:	Is it possible for a user in netwo	ork A to sub	scribe to a Voicemail box in
Comments	Oncok.	network B?	71 to 30b	Scribe to a voiceman box in
	Check:	Is the Event header in the SUB	CRIBE set	to 'message-summary'?
	Check:	Is the Accept header in the SUE		
		message-summary'?		
	Check:	Is the Event header in the NOT	TFY is set to	'message-summary'?
	Repeat t	his test in reverse direction.		

Test case number	SS_mwi_002		
Test case group	SIP-SIP/Service/MWI		
Reference	4.7.2/[16]		
SELECTION EXPRESSION	[Network A] SE 39 AND [Network B] SE 39		
Test purpose	A new entry in the Voicemail box is indicated to the owner.		
	The Voicemail owner is in network A, his Voicemail box is located in network B. Ensure when a user calls user A and the call is not answered, the call is forwarded to the Voicemail box of user A in network B. Ensure that the user A is		
	notified by message waiting indication that there is a new message present in his		
	voicemail account.		
Configuration	Voicemail in network B		
	Voicemail owner in network A		
SIP Parameter	NOTIFY		
	Subscription-State: active;expires=[any value]		
	Event: message-summary		
	Content-Type: application/simple-message-summary		
	Messages-Waiting: yes		
	Message-Account: sip:userA@networkA (optional) Voice-Message: [any new value]/[any old value] (optional)		
Message flow	Voice-iviessage. [arry new value]/[arry old value] (optional)		
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'?		
	Check: Is the Content-Type header in the NOTIFY set to 'application/simple-		
	message-summary'? Check: Contains the MIME body the header 'Messages-Waiting' set to 'yes'?		
	Contains the MIME body the optional header 'Message-Account'?		
	Contains the MIME body the optional header 'Voice-Message'?		
	Repeat this test in reverse direction.		
<u> </u>	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	nn1	
Test case group		Service/CC	
Reference	4.5.4.3/[1	14]	
SELECTION EXPRESSION	[Network	A] SE 40 AND [Network B] SE 40	
Test purpose	Indicating of CCBS possible.		
	User A is	located in network A and user B is	s located in network B.
	Ensure w	when user A calls user B and user E	B is busy, the network B sends an
		n that CCBS is possible in the 486	• ·
Configuration	maioatio	Tarac Gobo to possible in the fee	Bady Here iniai responde.
	400:		
SIP Parameter	486:		
	Call-Info: <sip:ue-b>;purpose=call-completion;m=BS</sip:ue-b>		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
,		INVITE	→ ` '
	←	486 Busy Here	_
	_	ACK	→
Comments	Check:	The 486 final response contains 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	• • •
	Repeat th	his test in reverse direction.	

	Tana and		
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 network B.	in	
	Check: The Call-Info header contains the purpose parameter set to		
	'call-completion' and the m parameter set to 'NR'. Repeat this test in reverse direction.		

Test case number	SS_cc_003		
Test case group	SIP-SIP/Service/CC		
Reference	4.5.4.2/[14]		
SELECTION EXPRESSION	([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR		
	[Network B] SE 41)		
Test purpose	Invocation of CCBS or CCNR.		
	 User A is located in network A and user B is located in network B. Ensure when user A call user B and user B is busy, the indication that CCBS is possible is sent to the network A. when user A invokes CCBS, a SUBSCRIBE request is sent to the network B, the Event header is set to 'call-completion' and the m parameter in the Request line is set to 'BS'. Ensure when user A call user B and user B is free, the indication that CCNR is possible is sent to the network A. when user A invokes CCNR, a SUBSCRIBE request is sent to the network B, the Event header is set to 'call-completion' and the m parameter in the Request line is set to 'NR'. Ensure that the network B sends a NOTIFY request to network A to confirm that the request is in the Call completion queue at the terminating Application Server. 		
Configuration	the request is in the can completion queue at the terminating Application Server.		
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 indication	Interconnection Interface SIP (Network B) on 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 00)4	
Test case group	SIP-SIP/S	Service/CC	
Reference	4.5.4.3/[14	4]	
SELECTION EXPRESSION	([Network	Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR	
	[Network	B] SE 41)	
Test purpose	Invocatio	n of CCBS or CCNR unsuccessful; short term denial	
	User A is	located in network A and user B is located in network B.	
	network B	at user A invokes a CCBS or CCNR request to network B and the sis currently unable to process the request (e.g. the B-queue is full), a corarily Unavailable final response is sent.	
Configuration			
SIP Parameter		E sip:B-AS;m=BS or m=NR	
		<ue-a></ue-a>	
	To: <ue-b></ue-b>		
	Contact: <a-as></a-as>		
	Event:	call-completion	
Message flow SIP (Network A)		Interconnection Interface SIP (Network B)	
An indicatio	n whether	CCBS or CCNR is possible is sent by network B	
	-	SUBSCRIBE →	
		480 (Temporarily Unavailable)	
Comments		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	
	Chook	request or set to 'NR' in case of CCNR?	
		Is a 480 Temporarily Unavailable sent from network B indicates the	
	CCBS or CCNR request is unsuccessful e.g. CC queue is full? Repeat this test in reverse direction.		
	Intepeat in	is test in reverse direction.	

Test case number	SS_cc_005	
Test case group	SIP-SIP/Service/CC	
Reference	4.5.4.3/[14]	
SELECTION EXPRESSION	([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR [Network B] SE 41)	
Test purpose	Successful CC operation	
	User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS or CCNR request. • Ensure when the user B becomes available for CC recall, the CC recall procedure is started. The network B sends a NOTIFY request to network A and a state header is present in the message body set to 'ready'. • Ensure that the recall from user A to user B is successful. • Ensure that a CC revocation notification is dent to network A to indicate the subscription is terminated; the reason header is set to 'noresource'.	
Configuration		
SIP Parameter	NOTIFY sip:O-AS Event:call-completion Content-Type: application/call-completion state: ready NOTIFY sip:O-AS Event:call-completion Subscription-State: terminated; reason=noresource	
Message flow SIP (Network A) 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 INVITE ACK Apply post text souting	
Comments	Apply post test routine Check: Is a NOTIFY request is sent to network A and the Event header is set	
Comments	to 'call-completion' and the state header in the message body is set to 'ready'? Check: Is the recall from user A to user B is successful? Check: Is the CC revocation is performed after the 180 Ringing or the 200 OK INVITE was sent to user A?	
	Repeat this test in reverse direction.	

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

Test case number	SS_cc_007
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.2/[14]
SELECTION EXPRESSION	([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR
	[Network B] SE 41)
Test purpose	User A is unavailable while CC recall is performed.
	User A is located in network A and user B is located in network B. User A has
	successfully invoked a CCBS or CCNR request. User B is available for CC-recall and network B sends a CC-recall notification to network A.
	 Ensure that network A sends PUBLISH request to suspend the recall procedure.
	Ensure that network A sends PUBLISH request to resume the recall
	procedure if user A is available to complete the recall procedure.
	Ensure the network B sends a NOTIFY request to indicate the CC-recall
	procedure.
Configuration	procedure.
SIP Parameter	NOTIFY sip:O-AS
	Event:call-completion
	Content-Type: application/call-completion
	state: ready
	PUBLISH sip B-AS
	To: SIP 2
	Event: presence
	Content-Type: application/pidf+xml
	xml version="1.0" encoding="UTF-8"?
	<pre><pre><pre><pre><status></status></pre></pre></pre></pre>
	\bu310>01036u\bu310>
	PUBLISH sip B-AS
	To: SIP 2
	Event: presence
	Content-Type: application/pidf+xml
	<pre><?xml version="1.0" encoding="UTF-8"?></pre>
	<pre><pre><pre><pre></pre></pre></pre></pre>
	<status></status>
NA	<basic>open</basic>
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)
	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	CODO possible
SIP (Network A)	Interconnection Interface SIP (Network B)
On (Network A)	INVITE -
	← 486 Busy Here (REL)
	ACK →
Comments	Check: The 486 final response contains an encapsulated BICC/ISUP REL, the
	Cause value set to 17 or 34 and the Diagnostics set to 'CCBS
	possible'.
	Repeat this test in reverse direction.

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

7.1.6 Other PSTN services (SIP-I interworking)

7.1.6.1 User-to-User Signaling (UUS)

Test case number	SS uus 001
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 63
Test purpose	SIP-I support: Indicating of User-to-User service 1 implicit in initial INVITE
	request.
	BICC/ISUP - SIP-I interworking applies in the originating network User A is
	located in network A and user B is located in network B.
	Ensure when user A subscribed to the User-to-User service 1 implicit request
	calls user B an User-to-user Information parameter is present in the
	encapsulated IAM of the initial INVTE request.
Configuration	User A is subscribed to the User-to-User service 1 implicit request
SIP Parameter	INVITE:
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	IAM
	User-to-user Information
	User Information
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE (IAM) →
	Apply post test routine
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request?
	Check: Is a User-to-user Information parameter present in the encapsulated
	ISUP/BICC IAM?
	Repeat this test in reverse direction.

Test case number	SS_uus_002
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1, 6.5/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE
	63
Test purpose	SIP-I support: Indicating of User-to-User service 1 implicit response in 180.
	BIOC/IOLID OID I'' I'' I'' I'' I'' I'' I'' I'' I''
	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	User information
SIP (Network A)	Interconnection Interface SIP (Network B)
On (Network A)	INVITE (IAM)
	← 180 Ringing (ACM)
	Apply post test routine
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request?
	Check: Is a User-to-user Information parameter present in the encapsulated
	ISUP/BICC IAM?
	Check: Is an ISUP/BICC ACM encapsulated in the 180 response?
	Check: Is a User-to-user Information parameter present in the encapsulated
	ISUP/BICC ACM?
	Repeat this test in reverse direction.

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

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

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

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

Test case number	SS_uus_008
Test case group	SIP-SIP/SIP-I/UUS
Reference	5.4.3.2, 6.5, 7.1/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	SIP-I support: Indicating of User-to-User service 2 in initial INVITE request successful.
	BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B.
	Ensure when user A subscribed to the User-to-User service 2 calls user B an
	User-to-user Indicator parameter is present set to 'Request service 2', 'not essential' or 'essential' in the encapsulated IAM of the initial INVTE request. The User-to-User service is successful.
Configuration	User A is subscribed to the User-to-User service 2
	User B is subscribed to the User-to-User service 2
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM
	User-to-user Indicator Request
	service 2 not essential or 'essential'
	180 Content-Type: application/isup;version=itu-t92
	Content-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
Message flow	User Information
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE (IAM) →
	← 180 Ringing (AĆM)
	INFO (USR) → 200 OK INFO
	← 183 Session Progress (USR)
_	Apply post test routine
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request and the a User-to-user Indicator parameter is set to Is the Request service 2 'not essential' or 'essential'?
	Check: Is an ISUP/BICC ACM encapsulated in the 180 and the User-to-user
	Indicator parameter is set to 'Response', 'service 2 provided'? Check: Is an ISUP/BICC USR encapsulated in the INFO message sent from
	network A to network B containing an User-to-user Information parameter?
	Check: Is an ISUP/BICC USR encapsulated in the 183 response sent from network B to network A containing an User-to-user Information
	parameter? Repeat this test in reverse direction.
1	

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

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

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

Test case number	SS_uus_012
Test case group	SIP-SIP/SIP-I/UUS
Reference	5.4.3.2, 6.5, 7.1/[24]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE
	63
Test purpose	SIP-I support: Indicating of User-to-User service 3 in initial INVITE request successful.
	BICC/ISUP - SIP-I interworking applies in the originating network User A is
	located in network A and user B is located in network B.
	Ensure when user A subscribed to the User-to-User service 3 calls user B an
	User-to-user Indicator parameter is present set to 'Request service 3', 'not
	essential' or 'essential' in the encapsulated IAM of the initial INVTE request. The User-to-User service is successful.
Configuration	User A is subscribed to the User-to-User service 3
	User B is subscribed to the User-to-User service 3
SIP Parameter	INVITE:
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM
	User-to-user Indicator
	Request
	service 3
	not essential or 'essential'
	200 OK
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ANM
	User-to-user Indicator
	Response service 3 provided
	ositios o providos
	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)
	← 180 Ringing (ACM) ← 200 OK INVITE (ANM)
	ACK →
	INFO (USR) →
	€ 200 OK INFO
	← INFO (USR)
	200 OK INFO Apply post test routine
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request and
	the
	a User-to-user Indicator parameter is set to Is the Request service 3
	'not essential' or 'essential'?
	Check: Is an ISUP/BICC ANM encapsulated in the 200 OK INVITE and the
	User-to-user Indicator parameter is set to 'Response', 'service 3 provided'?
	Check: Is an ISUP/BICC USR encapsulated in the INFO message sent from
	network A to network B containing an User-to-user Information
	parameter?
	Check: Is an ISUP/BICC USR encapsulated in the INFO message sent from
	network B to network A containing an User-to-user Information
	parameter? Repeat this test in reverse direction.
	וויסףסעג נווס נססג ווו וסיסוסט מוופטנוטוו.

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
l set pai pees	200 OK response.
	BICC/ISUP - SIP-I interworking applies in the originating and terminating network
	User A is located in network A and user B is located in network B.
	Ensure when user A subscribed to the User-to-User service 3 not essential calls
	user B not subscribed to User-to-User service 3 the call is rejected by the
	network an User-to-user Indicator parameter is present set to 'Response',
	'service 3 not provided' in the encapsulated ANM of the 200 OK final response.
Configuration	User A is subscribed to the User-to-User service 3
	User B is not subscribed to the User-to-User service 3
SIP Parameter	INVITE:
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	IAM
	User-to-user Indicator
	Request
	service 3
	not essential
	200 OK
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ANM
	User-to-user Indicator
	Response
	service 3 not provided
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE (IAM) →
	← 180 Ringing (ACM)
	← 200 OK INVITE (ANM)
	ACK →
	Apply post test routine
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request?
	Check: Is a User-to-user Information parameter present in the encapsulated
	ISUP/BICC IAM set to 'Request', 'service 3' 'not essential'?
	Check: Is an ISUP/BICC ANM encapsulated in the 200 OK response?
	Check: Is an User-to-user Indicator parameter present set to 'Response', 'service 3 not provided' in the encapsulated ISUP/BICC ANM?
	Repeat this test in reverse direction.
	וויבףבמו נוווס נבטו ווו ובייבוטב עוובטנוטוו.

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.
	DICC/ICLID CID Linterway line and in the origination and termination and termination
	BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B.
	Ensure when user A subscribed to the User-to-User service 3 essential calls
	user B not subscribed to User-to-User service 3 the call is rejected by the
	network. A 500 Server Internal Error is sent and an encapsulated ISUP/BICC
	REL is present, the Cause value is set to #29 or #69.
Configuration	User A is subscribed to the User-to-User service 3
SIP Parameter	INVITE:
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	IAM
	User-to-user Indicator
	Request
	service 3
	essential
	500
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	REL
	Cause value
	#29 or #69
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE (IAM) →
	← 500 Server Internal Error (REL)
	ACK →
Comments	Apply post test routine Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request?
	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
	llocated in network A and user B is located in network B.
	Ensure when user A subscribed to the User-to-User service 3 user A is able to
	request the User-to-User service 3 while the session is established. The User-to-
	User service is successful.
Configuration	User A is subscribed to the User-to-User service 3
	User B is subscribed to the User-to-User service 3
SIP Parameter	INFO:
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required
	FAR
	Facility indicator
	user-to-user service
	User-to-user Indicator
	Request
	service 3
	not essential
	INFO:
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	FAA
	Facility indicator
	user-to-user service
	User-to-user Indicator Response
	service 3 provided
	Scrivice o provided
	INFO
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	USR
	User-to-user Information User Information
Message flow	USEI IIIIOIIIIatioii
SIP (Network A)	Interconnection Interface SIP (Network B)
,	A session is already established
	INFO (FAR) →
	€ 200 OK INFO
	← INFO (FAA)
	200 OK INFO →
	INFO (USR) →
	€ 200 OK INFO
	← INFO (USR)
	200 OK INFO →
	Apply post test routine

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

Test case group 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 63 Test purpose SIP-I support: Indicating of User-to-User service 3 during a session in established unsuccessful. BICC/ISUP - SIP-I interworking applies in the originating network User A is all 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 all request the User-to-User service 3 while the session is established. The service B is not subscribed to the User-to-User service 3 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 Indicator Request	is is ble to
SELECTION EXPRESSION ([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND 63	is is ble to
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 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 all request the User-to-User service 3 while the session is established. The service 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 User B is not subscribed to the User-to-User service 3 INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FAR Facility indicator user-to-user service User-to-user Indicator	is is ble to
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 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 al request the User-to-User service 3 while the session is established. The service 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 INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FAR Facility indicator user-to-user service User-to-user Indicator	is is ble to
established unsuccessful. BICC/ISUP - SIP-I interworking applies in the originating network User A 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 al request the User-to-User service 3 while the session is established. The service 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 INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FAR Facility indicator	is ble to
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 al request the User-to-User service 3 while the session is established. The service 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 INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FAR Facility indicator user-to-user service User-to-user Indicator	ble to
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 INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FAR Facility indicator user-to-user service User-to-user Indicator	
Configuration User A is subscribed to the User-to-User service 3 User B is not subscribed to the User-to-User service 3 INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FAR Facility indicator user-to-user service User-to-user Indicator	
SIP Parameter Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FAR Facility indicator user-to-user service User-to-user Indicator	
user-to-user service User-to-user Indicator	
service 3 not essential	
INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FRJ Facility indicator	
user-to-user service	
User-to-user Indicator Response	
service 3 not provided	
Message flow SIP (Network A) Interconnection Interface A session is already established INFO (FAR) SIP (Network I	В)
← 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 for Network A to Network B and the a User-to-user Indicator parameters.	
set to Is the Request service 3 'not essential'? Check: Is an ISUP/BICC FAA encapsulated in the INFO request sent for Network B to Network A and the User-to-user Indicator parameter set to 'Response', 'service 3 not provided'?	
Repeat this test in reverse direction.	ļ

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-Type: application/isup,version=itu-tez Content-Disposition: signal;handling=required
	IAM
	Access transport
	Calling party subaddress
	[any boundary name]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
, ,	INVITE(IAM) →
Comments	Establish a call from User A subscribed to the SUB supplementary service to
	user B
	Check: Is an ISUP/BICC IAM present in the initial INVITE request?
	Check: Is an ISUP/BICC ATP parameter present in the encapsulated IAM
	containing a Calling party subaddress?
	Repeat this test in reverse direction.

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

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

7.1.6.3 Terminal Portability (TP)

Test case number	SS_tp_001	
Test case group	SIP-SIP/SIP-I/TP	
Reference	5.4.3.2/[24]	
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 64	
Test purpose	SIP-I support. SUS and RES messages transferred in an INFO request. BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. A session is already established. Ensure that an INFO request is sent from Network A to Network B and an ISUP SUS message is encapsulated containing a Suspend/resume	
	indicator set to ISDN subscriber initiated. Ensure that an INFO request is sent	
	from Network A to Network B and an ISUP RES message is encapsulated	
	containing a Suspend/resume indicator set to ISDN subscriber initiated.	
Configuration	User A is subscribed to the Terminal Portability supplementary service	
SIP Parameter	INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required SUS Suspend/resume indicator ISDN subscriber initiated	
	INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required RES Suspend/resume indicator ISDN subscriber initiated	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B) A confirmed session already exists INFO(SUS) → 200 OK INFO	
	INFO(RES) 200 OK INFO Apply post test routine	
Comments	A session is already established	
	Check: Is an ISUP SUS message encapsulated in the INFO request and the Suspend/resume indicator set to 'ISDN subscriber initiated'?	
	Check: Is an ISUP RES message encapsulated in the INFO request and the Suspend/resume indicator set to 'ISDN subscriber initiated'?	
	Repeat this test in reverse direction.	

Test case number	SS_tp_002	
Test case group	SIP-SIP/SIP-I/TP	
Reference	5.4.3.2, 6.11.2, 6.11.2/[24]	
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 64	
Test purpose	SIP-I support. SUS message transferred in an INFO request call released.	
	BICC/ISUP - SIP-I interworking applies in the originating network User A is	
	located in network A and user B is located in network B. A session is already	
	established. Ensure that an INFO request is sent from Network A to Network B	
	and an ISUP SUS message is encapsulated containing a Suspend/resume	
	indicator set to ISDN subscriber initiated. Ensure that an BYE request is sent	
	from Network A to Network B and an ISUP REL message is encapsulated	
	containing a Cause value set to #102.	
Configuration	User A is subscribed to the Terminal Portability supplementary service	
SIP Parameter	INFO	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required SUS	
	Suspend/resume indicator	
	ISDN subscriber initiated	
	TODIY GUDGOTIDGI TITUUKGU	
	BYE	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	REL	
	Location	
	public network serving remote user	
	Cause value	
	102	
NA		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)	
SIF (Network A)	A confirmed session already exists	
	INFO(SUS)	
	€ 200 OK INFO	
	200 01(1111)	
	BYE(REL) →	
	← 200 OK BÝE	
Comments	A session is already established	
	Check: Is an ISUP SUS message encapsulated in the INFO request and the	
	Suspend/resume indicator set to ISDN 'subscriber initiated'?	
	Check: Is an ISUP REL message encapsulated in the BYE request and the	
	Cause value set to #102?	
	Repeat this test in reverse direction.	

7.2 Number Portability

Test case number	SS_NP_0	001	
Test case group	SIP-SIP/	NubP	
Reference	5.3, 5.4/[2	2]	
SELECTION EXPRESSION	[Network	A] SE 13	
Test purpose	Request	line in the INVITE contains the number portability indication.	
	the INVIT paramete	tempts to call user B ported to network B. Ensure that the userinfo in E contains a destination number in the global number format, an 'rn' er containing the Number Portability Routing Number in a global number th hex digits and optional the 'npdi' parameter.	
Configuration			
SIP Parameter	INVITE: Request line		
		<cc> <ndc> <sn>[;npdi][; rn=(Number portability routing number)] <hostname>;user = phone SIP/2.0</hostname></sn></ndc></cc>	
Message flow			
SIP (Network A)		Interconnection Interface SIP (Network B) INVITE →	
	Apply post test routine		
Comments	Check:	Is the URI in the userinfo of the Request line in a global number format?	
	Check:	Is the URI rn parameter containing the Number Portability Routing	
		Number in a global number format?	
	Check:	Is optional the URI parameter 'npdi' present?	
	Check:	Is the user parameter set to 'phone'?	
	Repeat th	nis test in reverse direction.	

Test case number	SS_NP_002	
Test case group	SIP-SIP/NubP	
Reference	5.3, 5.4/[2]	
SELECTION EXPRESSION	NOT [Network A] SE 13	
Test purpose	Request line in the INVITE without npdi parameter.	
	The Network A does not have a Number Portability database. User A attempts to	
	call user B ported to network B. Ensure that the userinfo in the INVITE contains	
	a destination number in a global number format and a npdi URI parameter is not	
	present.	
Configuration		
SIP Parameter	INVITE: Request line	
	sip: + <cc> <ndc> <sn>@<hostname>;user = phone SIP/2.0</hostname></sn></ndc></cc>	
Message flow	1	
SIP (Network A)	Interconnection Interface SIP (Network B)	
	INVITE ->	
	Apply post test routine	
Comments	Check: Is the URI in the userinfo of the Request line in a global number format	
	without npdi parameter and number portability routing number?	
	Check: Is the user parameter set to 'phone'?	
	Repeat this test in reverse direction.	

7.3 Accounting

Test case number	SS_acc_001
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	Comparison of Charging Data Records > 1 s.
	Accounting of a confirmed session with a duration > 1 s. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.
Configuration	
SIP Parameter	
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → 180 Ringing 200 OK INVITE ACK → Communication BYE → 200 OK BYE
Comments	 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 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_002
Test case group	SIP-SIP/ACCOUNTING
Reference	OII -OII /ACCOUNTING
SELECTION EXPRESSION	O (OL D. (D L d.
Test purpose	Comparison of Charging Data Records < 1 s
	Accounting of a confirmed session with a duration of < 1 min. Verify the duration
	of the active session stored in the CDR of both networks compared with the
	duration in the monitored message flow at the Interconnection Interface.
Configuration	
SIP Parameter	
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE ->
	← 180 Ringing
	← 200 OK INVITE
	ACK →
	Communication
	BYE →
	← 200 OK BYE
Comments	Setup a call from network A to network B.
	Verify is the session confirmed.
	3. Terminate the session after 5 s.
	4. Determine the duration of the session from the trace of the call monitor.
	5. Compare the following information elements indicated in the CDR's of both
	networks:
	calling party number
	called party number
	timestamp
	callduration
	callsetuptime (optional)
	6. Check the duration indicated in the CDR against the duration in the call
	trace.
	7. Repeat this test in reverse direction.

Test case number	SS_acc_003
Test case group	SIP-SIP/ACCOUNTING
Reference	OII -OII /AGGGGITTING
SELECTION EXPRESSION	
Test purpose	Comparison of Charging Data Records > 15 min.
	Accounting of a confirmed session with a duration of > 15 min. Verify the
	duration of the active session stored in the CDR of both networks compared with
	the duration in the monitored message flow at the Interconnection Interface.
Configuration	the duration in the monitored message now at the interconnection interface.
SIP Parameter	
Message flow	Interconnection Interfere
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE ->
	180 Ringing
	← 200 OK INVITE
	ACK →
	Communication
	BYE →
	← 200 OK BYE
Comments	Setup a call from network A to network B.
	Verify is the session confirmed.
	3. Terminate the session after 15 min.
	4. Determine the duration of the session from the trace of the call monitor.
	5. Compare the following information elements indicated in the CDR's of both
	networks:
	calling party number
	called party number
	timestamp
	callduration
	callsetuptime (optional)
	6. Check the duration indicated in the CDR against the duration in the call
	trace.
	7. Repeat this test in reverse direction.

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

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

Test case number	100 000	
	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.	
	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	
	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	100 007		
	SS_acc_007		
Test case group	SIP-SIP/ACCOUNTING		
Reference			
SELECTION EXPRESSION			
Test purpose	Comparison of Charging Data Records more than 24 hours.		
	Accounting of a confirmed session with duration > 24 h with change of date. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.		
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE →		
	← 180 Ringing		
	← 200 OK IŇVIŤE		
	ACK →		
	Communication		
	BYE →		
	← 200 OK BYE		
Comments	Setup a call from network A to network B.		
	2. Verify is the session confirmed.		
	3. Terminate the session after 24 hours.		
	4. Determine the duration of the session from the trace of the call monitor.		
	5. Compare the following information elements indicated in the CDR's of both		
	networks:		
	calling party number		
	called party number		
	timestamp		
	• callduration		
	callsetuptime (optional)		
	6. Check the duration indicated in the CDR against the duration in the call		
	trace.		
	7. Repeat this test in reverse direction.		
	7. Repeat this test in reverse unection.		

Test case number	SS_acc_008		
Test case group	SIP-SIP/ACCOUNTING		
Reference			
SELECTION EXPRESSION			
Test purpose	Comparison of Charging Data Records less than 1 s.		
	Accounting of a confirmed session with duration <1 s. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.		
Configuration			
SIP Parameter			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE → 180 Ringing ← 200 OK INVITE ACK → Communication BYE → 200 OK BYE		
Comments	 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 callsetuptime (optional) Check the duration indicated in the CDR against the duration in the call trace. Repeat this test in reverse direction. 		

7.4 Carrier Selection

Test case number	SS_csel_001	
Test case group	SIP-SIP/CS	
Reference	5.7.1.10/[2]	
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15	
Test purpose	User selects an operator 'call-by-call'.	
root pan pooc		
	User A and user B are located in network A. Ensure that user A is able to call	
	user B and user A is able to select network B as a selected carrier 'call-by-call'.	
Configuration	User in network A is not presubscribed	
SIP Parameter	INVITE: Request line	
	sip: + <cc> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc></cc>	
	7	
	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 'cic' tel uri parameter present in the Request URI in the INVITE	
	sent from network A to network B identifying the selected carrier?	
	Check: Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A?	
	Check: Is optional the 'rn' parameter present in the Request URI of the INVITE	
	request sent from network B to network A?	
	NOTE 1: The 'cic' parameter may be absent according national regulations or	
	national agreements.	
	NOTE 2: It is possible that further informations are available in the Request line	
	regarding the end user charging in case of Carrier selection.	
	Repeat this test in reverse direction.	

Test case number	SS_csel_002
Test case group	SIP-SIP/CS
Reference	5.7.1.10/[2]
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15
Test purpose	User is presubscribed to operator B.
l cot pui pooc	occi io procussorissa to operator si
	User A and user B are located in network A. Ensure that user A is able to call
	user B and user A is preselected to network B as a selected carrier.
Configuration	User in network A is presubscribed to network B
SIP Parameter	INVITE: Request line
	sip: + <cc> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc></cc>
	INVITE: Request line
	sip: + <cc> <ndc> <sn>;npdi</sn></ndc></cc>
	[;rn= <number number="" portability="" routing="">]@<hostname>;</hostname></number>
	user=phone SIP/2.0
Message flow	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE 1 →
	INVITE 1 → (INVITE 2
SIP (Network A)	INVITE 1 → INVITE 2 Apply post test routine
	INVITE 1 → INVITE 2 Apply post test routine Check: Is the 'cic' tel uri parameter present in the Request URI in the INVITE
SIP (Network A)	INVITE 1 INVITE 2 Apply post test routine Check: Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier?
SIP (Network A)	INVITE 1 Apply post test routine Check: Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? Check: Is the 'npdi' parameter present in the Request URI of the INVITE
SIP (Network A)	INVITE 1 Apply post test routine Check: Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? Check: Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A?
SIP (Network A)	 INVITE 1 INVITE 2 Apply post test routine Check: Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? Check: Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? Check: Is optional the 'rn' parameter present in the Request URI of the INVITE
SIP (Network A)	INVITE 1 Apply post test routine Check: Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? Check: Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? Check: Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A?
SIP (Network A)	 Apply post test routine Check: Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? Check: Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? Check: Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? NOTE 1: The 'cic' parameter may be absent according national regulations or
SIP (Network A)	 Apply post test routine Check: Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? Check: Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? Check: Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? NOTE 1: The 'cic' parameter may be absent according national regulations or national agreements.
SIP (Network A)	 Check: Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? Check: Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? Check: Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? NOTE 1: The 'cic' parameter may be absent according national regulations or national agreements. NOTE 2: It is possible that further informations are available in the Request line
SIP (Network A)	 Apply post test routine Check: Is the 'cic' tel uri parameter present in the Request URI in the INVITE sent from network A to network B identifying the selected carrier? Check: Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A? Check: Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A? NOTE 1: The 'cic' parameter may be absent according national regulations or national agreements.

Test case number	SS_csel_0	003
Test case group	SIP-SIP/C	S
Reference	5.7.1.10/[2	
SELECTION EXPRESSION	[Network A	A] SE14 AND [Network B] SE15
Test purpose	User is pr	esubscribed to an operator unequal to B, and overrides the
	preselecti	ion with call-by-call via operator B.
		d user B are located in network A. User A is preselected to a network
		network B. Ensure that user A is able to call user B and user A is able
		etwork B as a selected carrier 'call-by-call'. The preselected carrier is
	ignored.	
Configuration		twork A is presubscribed to network B
SIP Parameter		equest 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>
		equest line
	sip	: + <cc> <ndc> <sn>;npdi</sn></ndc></cc>
		[;rn= <number number="" portability="" routing="">]@<hostname>;</hostname></number>
Manager		user=phone SIP/2.0
Message flow		Interconnection Interface SIP (Network B)
SIP (Network A)		Interconnection Interface SIP (Network B) INVITE 1 →
	←	INVITE 1
		Apply post test routine
Comments	Check:	Is the 'cic' tel uri parameter present in the Request URI in the INVITE
Comments		sent from network A to network B identifying the selected carrier?
		Is the 'npdi' parameter present in the Request URI of the INVITE
		request sent from network B to network A?
		Is optional the 'rn' parameter present in the Request URI of the INVITE
		request sent from network B to network A?
		The 'cic' parameter may be absent according national regulations or
		national agreements.
	NOTE 2:	It is possible that further informations are available in the Request line
		regarding the end user charging in case of Carrier selection.
	Repeat thi	s test in reverse direction.
,		

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

Test case number	SS csel	005	
Test case group	SIP-SIP/CS		
Reference			
SELECTION EXPRESSION	[Network	A] SE14 AND [Network B] SE15 AND [Network A] SE34	
Test purpose	User is p	preselected to operator B. Transit of CUG information -OA. That ing user in a CUG Outgoing Access not allowed preselected to B and calls to a user in the same CUG. The session establishment is	
Configuration	User in n	network A is presubscribed to network B network A are in the same CUG	
SIP Parameter	INVITE: Request line sip: + <cc> <ndc> <sn>@tariff.<hostname> user=phone SIP/2.0 Content-Type: application/vnd.etsi.cug+xml</hostname></sn></ndc></cc>		
	c	eug> :: cugCommunicationIndicator>11 : cugCommunicationIndicator	
	si Conte	Request line p: + <cc> <ndc> <sn@<hostname>;user=phone SIP/2.0 ent-Type: application/vnd.etsi.cug+xml ent-Disposition:;handling= required</sn@<hostname></ndc></cc>	
	<:cug> <: cugCommunicationIndicator>11 : cugCommunicationIndicator		
Manager flam	<:C	rug>	
Message flow SIP (Network A)	+	Interconnection Interface SIP (Network B) INVITE 1 INVITE 2 Apply post test routine	
Comments	Check:	Is the sub domain pattern 'tariff' present at the beginning of the hostportion only of the initial INVITE sent from network A to network B?	
	Check:	Is the 'npdi' parameter present in the userinfo of the INVITE request sent from network B to network A? Is optional the 'rn' parameter present in the userinfo of the INVITE	
	Check:	request sent from network B to network A? Contains the XML body in the INVITE a 'cugCommunicationIndicator'	
	Check:	element set to '11' as a 'cug' child element? Is the session setup not rejected?	

7.5 Emergency call

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

7.6 SIP Support of Charging

Test case number	SS_sipc_	001
Test case group		SIP_charging
Reference	B.2.3/[19]
SELECTION EXPRESSION	SE 16	
Test purpose	User A is in case o information	Iful session from user A to user B via network B one single tariff. I 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 all or successful final response.
Configuration		
SIP Parameter	INVITE:	
oir Falametei	Supp 18x or 20 Requ Conte Conte messa crgi ch	orted: 100rel 00 OK ire: 100rel entType: application/vnd.etsi.sci+xml ent-Disposition: render; handling=optional ageType argingControlIndicators argingTariff tariffCurrency currentTariffCurrency currencyFactorScale currencyFactor currencyScale tariffDuration subTariffControl tariffControlIndicators iginationIdentification irrency (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?
	Chack	Is the tariffCurrency element set to 'currentTariffCurrency'?
	Check:	
	Check:	Represents the currencyFactorScale in the
		communicationChargeSequenceCurrency element the applicable
	Oh!-	tariff?
	Check:	Is the tariffDuration element set to '0'?
	Check:	Is the optional element 'currency' set to 'EUR' if present?
	Repeat th	nis test in reverse direction.

Test case number	SS_sipc_	002		
Test case group		SIP_charging		
Reference	B.2.3/[19			
SELECTION EXPRESSION	SE 16	1		
Test purpose		Iful session from user A to user B via network B several tariffs in		
rest purpose	one seq	uence.		
	in case o	s located in network A and network B is responsible for charging (CDP) of carrier selection or service. Ensure that the network B sends a tariff		
		on with several tariffs in a sequence covered in a XML MIME body in a provisional or successful final response.		
Configuration				
SIP Parameter	INVITE: Supp	orted: 100rel		
	18x or 20	UU OK		
		ire: 100rel		
	Conte	entType: application/vnd.etsi.sci+xml ent-Disposition: render; handling=optional		
	mess crgt	ageType		
	ch	nargingControlIndicators		
	ch	nargingTariff		
		tariffCurrency		
		currentTariffCurrency communicationChargeSequenceCurrency		
		currencyFactorScale		
		currencyFactor		
		currencyScale		
		tariffDuration		
		subTariffControl		
		communicationChargeSequenceCurrency		
		currencyFactorScale		
		currencyFactor		
		currencyScale tariffDuration		
		subTariffControl		
		tariffControlIndicators		
	01	riginationIdentification		
		urrency (optional)		
Message flow SIP (Network A)		Interconnection Interface SIP (Network B)		
	-	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		
	Check:	INVITE final response? Is the tariffCurrency element set to 'currentTariffCurrency'?		
		Are there more than one communicationCharge		
	OHECK.	SequenceCurrency elements present in the currentTariffCurrency		
	01 1	element?		
	Cneck:	Represents the currencyFactorScale in the communicationCharge SequenceCurrency elements the applicable tariffs?		
	Check:	Is the tariffDuration element in the last applicable tariff set to '0'?		
		Is the optional element ' currency ' set to 'EUR' if present?		
		his test in reverse direction.		
	1 6			

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		
Managaraflaw	currency (optional)		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →		
CASE A	← 18x(crgt) PRACK →		
CASE B	← 200 OK PRACK ← 200 OK INVITE(crat)		
	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 'cell'AttemptChargeCurrency'?		
	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	Providence of outcomes in an income of the company		
SIP Parameter	INVITE: Supported: 100rel		
	18x or 200 OK Require: 100rel ContentType: application/vnd.etsi.sci+xml Content-Disposition: render; handling=optional		
	messageType crgt chargingControlIndicators chargingTariff tariffCurrency		
	currentTariffCurrency communicationChargeSequenceCurrency currencyFactorScale		
	currencyFactor currencyScale tariffDuration subTariffControl		
	tariffControlIndicators callSetupChargeCurrency currencyFactor		
	currencyScale		
	originationIdentification currency (optional)		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →		
CASE A	← 18x(crgt) PRACK → 200 OK PRACK		
CASE B	200 OK INVITE(crgt) Apply post test routine		
Comments	Check: Is the supported header in the initial INVITE set to '100rel'? Check: Is the Require header in the response containing the tariff information set to '100rel'?		
	Check: Is the messageType a ' crgt ' present in a 1xx provisional or a 200 OK INVITE final response?		
	Check: Is the tariffCurrency element set to 'callSetupChargeCurrency'? Check: Represents the currencyFactorScale in the callSetupChargeCurrency element the applicable tariff? Check: Is the optional element 'currency' set to 'EUR' if present?		
	Repeat this test in reverse direction.		

Test case number	SS_sipc_	005	
Test case group		SIP_charging	
Reference	B.2.3/[19		
SELECTION EXPRESSION	SE 16	1	
Test purpose		ful session from user A to use	r B via network B with a next tariff.
	User A is in case of information	s located in network A and networ of carrier selection or service. Ens	k B is responsible for charging (CDP) ure that the network B sends a tariff ch over time covered in a XML MIME
Configuration			
SIP Parameter	18x or 200 Requi Conte Conte messa crgi ch	re: 100rel entType: application/vnd.etsi.sci+xml ent-Disposition: render; handling=option ageType margingControlIndicators margingTariff tariffCurrency currentTariffCurrency currencyFactorScale currencyFactor currencyScale tariffDuration subTariffControl tariffControlIndicators tariffSwitchCurrency communicationChargeSequentiffControl tariffControlIndicators tariffSwitchCurrency communicationChargeSequentiffControl tariffControlIndicators tariffSwitchCurrency communicationChargeSequentiffControl tariffControlIndicators tariffControlIndicators tariffControlIndicators tariffControlIndicators tariffControlIndicators tariffControlIndicators tariffControlIndicators tariffControlIndicators tariffSwitchOverTime	uenceCurrency SequenceCurrency le
Message flow		urrency (optional)	
SIP (Network A)		Interconnection Interface INVITE	SIP (Network B)
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 in	nitial INVITE set to '100rel'?
	Check: Check: Check: Check:	Is the Require header in the resset to '100rel'? Is the messageType 'crgt' preseINVITE final response? Is the tariffSwitchCurrency eleRepresents the currencyFacto communicationChargeSequence	ponse containing the tariff information ent in a 1xx provisional or a 200 OK ement set to 'nextTariffCurrency'?
	Check:	element? Is the optional element 'currenchis test in reverse direction.	

Test case number	SS sinc 006	
Test case group	SS_sipc_006 SIP-SIP/ SIP_charging	
Reference		
	B.2.3/[19] SE 16	
SELECTION EXPRESSION		
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	
	chargingControlIndicators chargingTariff tariffCurrency currentTariffCurrency communicationChargeSequenceCurrency currencyFactor currencyFactor currencyScale tariffDuration subTariffControl communicationChargeSequenceCurrency currencyFactorScale currencyFactors currencyFactor currencyFactor currencyScale tariffDuration subTariffControl tariffControlIndicators tariffSwitchCurrency nextTariffCurrency communicationChargeSequenceCurrency currencyFactor currencyFactor currencyFactor currencyFactor currencyFactor currencyFactor currencyFactor currencyScale tariffDuration subTariffControl communicationChargeSequenceCurrency currencyFactor	
Message flow	1 2 2 3 (1 2 2 2)	
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?	
Comments	Check: Is the tariffCurrency element set to 'currentTariffCurrency'? Check: Represents the currencyFactorScale in the communicationChargeSequenceCurrency elements the current tariffs? Check: Represents the currencyFactorScale in the communicationChargeSequenceCurrency elements the next tariffs?	

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

7.7 Quality of Service

7.7.1 Reference Configurations

7.7.1.1 Backbone Configuration

Figure 7.7-1 shows the backbone configuration.



Figure 7.7-1: Backbone

7.7.1.2 PSTN/ISDN classic access Configuration

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

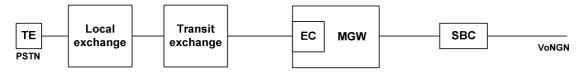


Figure 7.7-2: Reference configuration for PSTN/ISDN with classical access

7.7.1.3 NGN PSTN/ISDN access Configuration

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

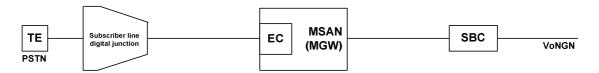


Figure 7.7-3: Reference configuration for NGN with PSTN/ISDN access

7.7.1.4 Access DSL Configuration

Figure 7.7-4 shows the xDSL access configuration.

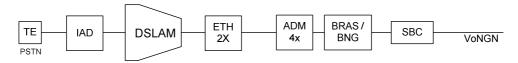


Figure 7.7-4: Reference configuration for DSL access

7.7.1.5 Delay Values

The requirements for the backbone delay, Network parameters: End-to-End Delay, Talker Echo Loudness Rating, R Value Delay with regional propagation delay ($1\,400\,\text{km}/11\,\text{ms}$) 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 Dtr seg A-B = (Dsum seg A-B- DJB1seg B- DJB2segA)/2 Loop in user segment A Dtr seg B-A = (Dsum seg B-A - DJB1seg B- DJB2segA)/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".
- ITU-T Recommendation Q.951: "Stage 3 description for number identification supplementary services using DSS 1".
- ITU-T Recommendation Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".
- ITU-T Recommendation 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	