ETSI TS 101 585 V2.1.1 (2018-02)



Core Network and Interoperability Testing (INT); IMS interconnection tests at the Ic Interface; (3GPP[™] Release 13); Test Suite Structure and Test Purposes (TSS&TP) Reference RTS/INT-00134

Keywords

IMS, interconnection, PSTN, SIP, testing, TSS&TP, UNI

ETSI

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

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

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

Important notice

The present document can be downloaded from: <u>http://www.etsi.org/standards-search</u>

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

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other ETSI documents is available at <u>https://portal.etsi.org/TB/ETSIDeliverableStatus.aspx</u>

If you find errors in the present document, please send your comment to one of the following services: https://portal.etsi.org/People/CommiteeSupportStaff.aspx

Copyright Notification

No part may be reproduced or utilized in any form or by any means, electronic or mechanical, including photocopying and microfilm except as authorized by written permission of ETSI. The content of the PDF version shall not be modified without the written authorization of ETSI. The copyright and the foregoing restriction extend to reproduction in all media.

> © ETSI 2018. All rights reserved.

DECT[™], PLUGTESTS[™], UMTS[™] and the ETSI logo are trademarks of ETSI registered for the benefit of its Members. **3GPP**[™] and LTE[™] are trademarks of ETSI registered for the benefit of its Members and of the 3GPP Organizational Partners. **oneM2M** logo is protected for the benefit of its Members.

GSM® and the GSM logo are trademarks registered and owned by the GSM Association.

Contents

Intelle	ectual Property Rights	5
Forew	ord	5
Moda	l verbs terminology	5
1	Scope	6
2	References	6
2.1	Normative references	6
2.2	Informative references	8
3	Definitions, conventions and abbreviations	9
3.1	Definitions.	
3.2	Conventions for representation of SIP/SDP information	
3.3	Abbreviations	
4	Test Suite Structure (TSS)	12
5	Declarations	12
5 5.1	Numbering Scheme	
5.2	Reference configuration	
5.3	Selection of End Devices	
6	Selection Expressions	
7	Test purposes	
, 7.0	Introduction	
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	
7.1.5.2	I I	
7.1.5.4	· ·	
7.1.5.5	· ·	
7.1.5.6		
7.1.5.6		
7.1.5.6		114
7.1.5.6		
7.1.5.6	e ee v	
7.1.5.6		
7.1.5.6	1	
7.1.5.7		
7.1.5.9		
7.1.5.1		
7.1.5.1		
7.1.5.1	2 Malicious Communication Identification (MCID)	210
7.1.5.1		214
7.1.5.1		. .
714	No Reply (CCNR)	
7.1.6	Other PSTN services (SIP-I interworking)	
7.1.6.1		
7.1.6.2	8()	
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 (optional)	
7.7.1	Delay Values	
7.7.2	Test purposes for Quality of Service test	
Anne	ex A (informative): Test case list and test case selection	
A.0	Introduction	
A.1	First step - "Identification of the Networks"	
A.2	Second step - Selection Expression	
A.3	Third step - Access and End Devises Types	
A.4	Fourth step - activation	
Anne	ex R (informative). Bibliography	274

Annex B (informative):	Bibliography	274
History		276

Intellectual Property Rights

Essential patents

IPRs essential or potentially essential to normative deliverables 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 (https://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.

Trademarks

The present document may include trademarks and/or tradenames which are asserted and/or registered by their owners. ETSI claims no ownership of these except for any which are indicated as being the property of ETSI, and conveys no right to use or reproduce any trademark and/or tradename. Mention of those trademarks in the present document does not constitute an endorsement by ETSI of products, services or organizations associated with those trademarks.

Foreword

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

Modal verbs terminology

In the present document "shall", "shall not", "should", "should not", "may", "need not", "will", "will not", "can" and "cannot" are to be interpreted as described in clause 3.2 of the <u>ETSI Drafting Rules</u> (Verbal forms for the expression of provisions).

"must" and "must not" are NOT allowed in ETSI deliverables except when used in direct citation.

1 Scope

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

2 References

2.1 Normative 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 referenced document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found at https://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.

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

[1]	ETSI TS 129 165: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Inter-IMS Network to Network Interface (NNI) (3GPP TS 29.165 Release 13)".
[2]	ETSI TS 124 229: "Digital cellular telecommunications system (Phase 2+) (GSM); 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 13)".
[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 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+) (GSM); 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 13)".
[8]	ETSI TS 124 608: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.608 Release 13)".
[9]	ETSI TS 124 604: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Communication Diversion (CDIV) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.604 Release 13)".
[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 13)".

- [11] ETSI TS 124 629: "Digital cellular telecommunications system (Phase 2+) (GSM); 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 13)".
- [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 13)".
- [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 13)".
- [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 13)".
- [15] ETSI TS 124 615: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification (3GPP TS 24.615 Release 13)".
- [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 13)".
- [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 13)".
- [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 13)".
- [19] ETSI TS 129 658: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; SIP Transfer of IP Multimedia Service Tariff Information; Protocol specification (3GPP TS 29.658 Release 13)".
- [20] ETSI TS 124 628: "Digital cellular telecommunications system (Phase 2+) (GSM); 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 13)".
- [21] IETF RFC 5009 (September 2007): "Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media".
- [22] Recommendation ITU-T V.152 (09-2010): "Procedures for supporting voice-band data over IP networks".
- [23] Recommendation ITU-T T.38 (11-2015): "Procedures for real-time Group 3 facsimile communication over IP networks".
- [24] Recommendation ITU-T Q.1912.5: "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".
- [25] ETSI TS 183 036: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ISDN/SIP interworking; Protocol specification".
- [26] IETF RFC 4733: "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals".

- [27] IETF RFC 4028: "Session Timers in the Session Initiation Protocol (SIP)".
- [28] Recommendation ITU-T Q.4016 (08-2016): "Testing specification of call establishment procedures based on SIP/SDP and ITU-T H.248 for a real-time fax over IP service".
- [29] ETSI TS 101 563 (V1.3.1): "Speech and multimedia Transmission Quality (STQ); IMS/PES/VoLTE exchange performance requirements".
- [30] ETSI ES 202 765-2 (V1.2.1): "Speech and multimedia Transmission Quality (STQ); QoS and network performance metrics and measurement methods; Part 2: Transmission Quality Indicator combining Voice Quality Metrics".
- [31] Recommendation ITU-T Q.543: "Digital exchange performance design objectives".
- [32] ETSI TS 102 250-2: "Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in mobile networks; Part 2: Definition of Quality of Service parameters and their computation".

2.2 Informative 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 referenced document (including any amendments) applies.

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

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

[i.1]	ETSI EN 300 403-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1) protocol; Signalling network layer for circuit-mode basic call control; Part 1: Protocol specification [ITU-T Recommendation Q.931 (1993), modified]".
[i.2]	ISO/IEC 9646 (1994): "Information technology Open Systems Interconnection Conformance testing methodology and framework".
[i.3]	ETSI TR 102 775 (V1.5.1): "Speech and multimedia Transmission Quality (STQ); Guidance on objectives for Quality related Parameters at VoIP Segment-Connection Points; A support to NGN transmission planners".
[i.4]	Recommendation ITU-T Q.1902.2 (07-2001): "Bearer Independent Call Control protocol (Capability Set 2) and Signalling System No.7 ISDN User Part: General functions of messages and parameters".
[i.5]	Recommendation ITU-T G.826: "End-to-end error performance parameters and objectives for international, constant bit-rate digital paths and connections".
[i.6]	ETSI TS 183 043: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation; Stage 3 specification".
[i.7]	Recommendation ITUT-T Q.76: "Service procedures for Universal Personal Telecommunication - Functional modelling and information flows".
[i.8]	Recommendation ITUT-T Q.764: "Signalling System No. 7 - ISDN User Part signalling procedures".
[i.9]	ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+) (GSM); 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)".

3 Definitions, conventions and abbreviations

3.1 Definitions

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

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 IETF RFC 3261 [4].

Basic Call Control (BCC): signalling protocol associated with the DSS1 - ISDN Basic Call control procedures of ETSI EN 300 403-1 [i.1]

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

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

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

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: mechanism for reserving bearer resources that is required for certain access technologies

NOTE: As defined in IETF RFC 3312 [6].

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].

3.2 Conventions for representation of SIP/SDP information

1) All letters of SIP method names are in capital.

EXAMPLE 1: INVITE, INFO.

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

EXAMPLE 2: To, From, Call-ID.

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/IETF RFC 3261 [4]), in plain text without surrounding angle brackets.

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

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

EXAMPLE 4: sip: URI, tel: URI.

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

EXAMPLE 5: 100 Trying, 484 Address Incomplete.

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

10

EXAMPLE 6: 200 OK INVITE, 200 OK PRACK.

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

EXAMPLE 7: m=line, a=line.

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

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

3.3 Abbreviations

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

ACR	Anonymous Communication Rejection
ACR-CB	Anonymous Call Rejection and Call Barring
AS	Application Server
ATP	Access Transport Parameter
ATS	Abstract Test Suite
BC	Bearer Capability
BCALL	Basic CALL
CB	Communication Barring
CC	Completion of Completion
CCBS	Completion of Computition Completion of Communications to Busy Subscriber
CCNR	Completion of Communications to Busy Subscriber
CD	Communication Deflection
CDIV	Communication Diversion
CDP	Charging Determinating Point
CDR	Communication Data Record
CFB	Communication Forwarding Busy
CFNL	Communication Forwarding Busy Communication Forwarding Not Logged in
CFNR	Communication Forwarding Not Logged In Communication Forwarding No Reply
CFU	Communication Forwarding Unconditional
CN	Core Network
COLP	Connected Line Presentation
	Connected Line Restriction
COLR CONF	Conference
CONF	Circuit Switched
CUG	Closed User Group
CUG-OA	Closed User Group Outgoing Access
CW	Communication Waiting
DTMF	Dual Tone Multi-Frequency
ECT	Explicit Communication Transfer
FFS	For Further Study
GSM	Global System for Mobile Communications
GW	GateWay
HLC	High layer Compatibility
HOLD	Communication Hold
IA	In coming Access allowed
IBCF	Interconnection Boarder Control Function
ICB	Incoming Call Barring
IMS	IP Multimedia core network Subsystem
IP	Internet Protocol
ISDN	Integrated Services Digital Network

IUTImplementation Under TestLLCLow Layer ComaptibilityMCIDMalicious Communication IdentificationMGCFMedia Gateway Control FunctionMWIMessage Waiting IndicationNDUBNetwork Determined User BusyOAOutgoing AccessOIPOriginating Identification Presentation RestrictionPBXPrivate Branch eXchagePDPProgrammable Data ProcessorPICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSESelection ExpressionSIP-ISIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest Suite StructureTSSTest Suite StructureTSSTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ServerUDUBUser Determined User BusyUEUser Equipment A sideUF-AUser Equipment B sideURUser Service InformationUUSUser to User ServiceXMLeXtensible Markup Language	ISUP	ISdn User Part
LLCLow Layer ComaptibilityMCIDMalicious Communication IdentificationMGCFMedia Gateway Control FunctionMWIMessage Waiting IndicationNDUBNetwork Determined User BusyOAOutgoing AccessOIPOriginating Identification PresentationOIROriginating Identification presentation RestrictionPBXPrivate Branch eXchagePDPProgrammable Data ProcessorPICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSESelection ExpressionSIPSession Initiation ProtocolSESelection ExpressionSIPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification RestrictionTRTerminating Identification RestrictionTRTerminating Identification RestrictionTRTerminating Identification RestrictionTRTest Suite StructureTSS&Test Suite StructureTSS&Test Suite StructureTSS&Test Suite StructureTSS&Test Suite StructureTSS&Test Suite StructureTSSTest Suite StructureTSSTest Suite StructureUACUser Agent <t< td=""><td></td><td>Implementation Under Test</td></t<>		Implementation Under Test
MCIDMalicious Communication IdentificationMGCFMedia Gateway Control FunctionMWIMessage Waiting IndicationMWIMessage Waiting IndicationNDUBNetwork Determined User BusyOAOutgoing AccessOIPOriginating Identification PresentationOIROriginating Identification presentation RestrictionPBXPrivate Branch eXchagePDPProgrammable Data ProcessorPICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSIPSession Initiation ProtocolSIP-1SIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification RestrictionTPTest Suite StructureTSS&Test Suite StructureTSS&Test Suite StructureTSS&Test Suite StructureUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUSIUser Service InformationUUSUser to User Service		
MGCFMedia Gateway Control FunctionMWIMessage Waiting IndicationNDUBNetwork Determined User BusyOAOutgoing AccessOIPOriginating Identification Presentation RestrictionPIRPrivate Branch eXchagePDPProgrammable Data ProcessorPICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSIPSession Initiation ProtocolSIPSession Initiation ProtocolSIPSupplementary SerciceSUBSUB addressingTIPTerminating Identification RestrictionTRTerminating Identification RestrictionTRTerst Surb StructureTSS&TPTest Suite StructureTSS&TPTest Suite StructureTSS&TPTest Suite StructureTSS&TPTest Suite StructureUAUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser Equipment A sideUE-AUser Equipment B sideURUniform Resource IdentifierUSIUser to User Service	MCID	
MWIMessage Waiting IndicationNDUBNetwork Determined User BusyOAOutgoing AccessOIPOriginating Identification PresentationOIROriginating Identification presentation RestrictionPBXPrivate Branch eXchagePDPProgrammable Data ProcessorPICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYPaTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSESupplementary SerciceSUBSUB addressingTIPTerminating Identification RestrictionTRTerminating Identification RestrictionTRTerminating Identification RestrictionTRTerminating Identification RestrictionTRTerminating Identification RestrictionTPTest Suite StructureTSSTest Suite StructureTSS&T Past Suite Structure and Test PurposesUAUser AgentUACUser Agent ServerUDUBUser Cetermined User BusyUEUser Equipment A sideUE-AUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service		Media Gateway Control Function
NDUBNetwork Determined User BusyOAOutgoing AccessOIPOriginating Identification PresentationOIROriginating Identification presentation RestrictionPBXPrivate Branch eXchagePDPProgrammable Data ProcessorPICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYPaTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSIPSession Initiation ProtocolSIP-ISIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification RestrictionTRTerminating Identification RestrictionTRTerminating Identification RestrictionTRTerst Suite StructureTSSTest Suite StructureTSS&TTest Suite StructureTSS&TTest Suite Structure and Test PurposesUAUser Agent ClientUASUser Agent ServerUDUBUser Equipment A sideUE-AUser Equipment B sideURIUniform Resource IdentifierUSIUser Service InformationUUSUser to User Service		
OAOutgoing AccessOIPOriginating Identification PresentationOIROriginating Identification presentation RestrictionPBXPrivate Branch eXchagePDPProgrammable Data ProcessorPICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSESupplementary SerciceSUBSUB addressingTIPTerminating Identification RestrictionTPTest Suite StructureTSSTest Suite StructureTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Capityment A sideUE=User Equipment B sideURIUniform Resource IdentifierUSIUser Service	NDUB	
OIPOriginating Identification PresentationOIROriginating Identification presentation RestrictionPBXPrivate Branch eXchagePDPProgrammable Data ProcessorPICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIP-ISIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest Suite StructureTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	OA	
OIROriginating Identification presentation RestrictionPBXPrivate Branch eXchagePDPProgrammable Data ProcessorPICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest Suite StructureTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	OIP	
PBXPrivate Branch eXchagePDPProgrammable Data ProcessorPICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSIPSession Initiation ProtocolSIPSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest Suite StructureTSS&Test Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser Service	OIR	
PDPProgrammable Data ProcessorPICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSIP-ISIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest Suite StructureTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser EquipmentUE-AUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	PBX	
PICSProtocol Implementation Conformance StatementPIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSIPSession Initiation ProtocolSIP-ISIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification RestrictionTRTerminating Identification RestrictionTPTest Suite StructureTSSTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	PDP	
PIXITProtocol Implementation eXrtra InformationPLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSIPSession Initiation ProtocolSIPSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest Suite StructureTSSTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	PICS	
PLMNPublic Land Mobile NetworkPSTNPublic Switched Telephone NetworkPTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSIPSession Initiation ProtocolSIPSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest Suite StructureTSS&Test Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service		
PTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSIPSession Initiation ProtocolSIPSubplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideURIUniform Resource IdentifierUSIUser to User Service	PLMN	
PTPayload TypePTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSIPSession Initiation ProtocolSIPSession Initiation ProtocolSIPSub endersingSIPSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	PSTN	Public Switched Telephone Network
PTYParTYQoSQuality of ServiceRTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSIPSession Initiation ProtocolSIP-1SIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	PT	*
RTPRealtime Transport ProtocolSDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSIP-1SIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser EquipmentUEUser EquipmentUE-AUser Equipment A sideURIUniform Resource IdentifierUSIUser to User Service	PTY	ParTY
SDPSession Description ProtocolSESelection ExpressionSIPSession Initiation ProtocolSIP-ISIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser EquipmentUEUser Equipment A sideUE-AUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	QoS	Quality of Service
SESelection ExpressionSIPSession Initiation ProtocolSIP-ISIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser EquipmentUEUser EquipmentUE-AUser Equipment A sideURIUniform Resource IdentifierUSIUser to User Service	RTP	Realtime Transport Protocol
SIPSession Initiation ProtocolSIP-ISIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideURIUniform Resource IdentifierUSIUser to User Service	SDP	Session Description Protocol
SIP-ISIP containing ISUPSSSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideURIUniform Resource IdentifierUSIUser to User Service	SE	Selection Expression
SSSupplementary SerciceSUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideURIUniform Resource IdentifierUSIUser to User Service	SIP	Session Initiation Protocol
SUBSUB addressingTIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Determined User BusyUEUser EquipmentUE-AUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	SIP-I	SIP containing ISUP
TIPTerminating Identification PresentationTIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Determined User BusyUEUser EquipmentUE-AUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	SS	Supplementary Sercice
TIRTerminating Identification RestrictionTPTest PurposeTSSTest Suite StructureTSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	SUB	
TPTest PurposeTSSTest Suite StructureTSSTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	TIP	Terminating Identification Presentation
TSSTest Suite StructureTSSTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	TIR	Terminating Identification Restriction
TSS&TPTest Suite Structure and Test PurposesUAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideURIUniform Resource IdentifierUSIUser to User Service		
UAUser AgentUACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser to User Service	TSS	Test Suite Structure
UACUser Agent ClientUASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser Service InformationUUSUser to User Service	TSS&TP	Test Suite Structure and Test Purposes
UASUser Agent ServerUDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser Service InformationUUSUser to User Service		
UDUBUser Determined User BusyUEUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser Service InformationUUSUser to User Service	UAC	
UEUser EquipmentUE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser Service InformationUUSUser to User Service	UAS	-
UE-AUser Equipment A sideUE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser Service InformationUUSUser to User Service		
UE-BUser Equipment B sideURIUniform Resource IdentifierUSIUser Service InformationUUSUser to User Service	-	
URIUniform Resource IdentifierUSIUser Service InformationUUSUser to User Service		
USIUser Service InformationUUSUser to User Service		
UUS User to User Service		
XMLeXtensible Markup Language		
	XML	eXtensible Markup Language

4 Test Suite Struct	ture (TSS)
---------------------	------------

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

5 Declarations

5.1 Numbering Scheme

FFS.

5.2 Reference configuration

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

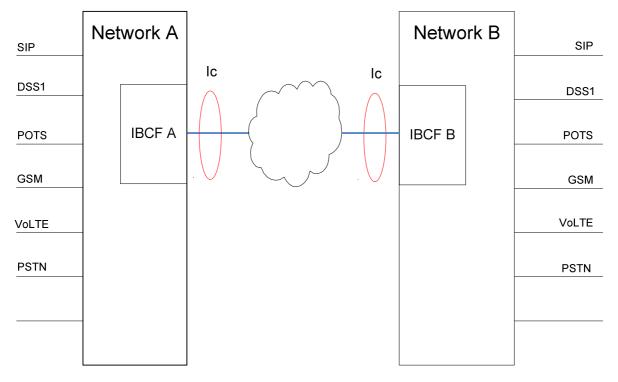


Figure 5.2-1: Reference configuration for the interconnection test

5.3 Selection of End Devices

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

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

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

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

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

Table 5.3-1: Overview of end devices

ETSI

6 Selection Expressions

Table 6-1 is used to select the optional Test Purposes for the compatibly test between network operator A and network operator B.

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

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

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

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

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

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

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

	SELECTION EXPRESSION:	Support	Support
		Network A	Network B
SE 28:	The Network supports Communication Forwarding Not Logged in		
	(CFNL)?		
SE 29:	The Network supports Communication Deflection?		
SE 30:	The Network supports the CDIV Notification procedure?		
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)?		
	Terminal capabilities		
SE 42:	The End device requires the resource reservation?		
SE 43:	The End device supports Fax transmission via G.711 codec?		
SE 44:	The End device supports Fax transmission via V.152 codec?		
SE 45:	The End device supports Fax transmission via m-line T.38 codec?		
SE 46:	A SIP end device is used supporting an ISDN user equipment and the PSTN XML Schema is used?		
SE 47:	End device is located in the PSTN or PLMN?		
SE 48:	The terminating UE supports the from-change tag procedure and sends a second user identity in an UPDATE request after the dialogue is confirmed?		
SE 49:	The end device performs ECT using the 'Blind/assured transfer'?		
SE 50:	The end device performs ECT using the 'Dind/assured transfer'?		
SE 50. SE 51:	The end device performs LOT dsing the consultative transfer ? The end device supports the Resource reservation procedure?		
<u>5L 51.</u>	PSTN/PLMN Supplementary services		
SE 52:	CLIP/CLIR is supported in the PSTN/PLMN part of the network?		
SE 52A:	The network supports the "Special arrangement" procedure for the originating user?		
SE 53:	COLP/COLR is supported in the PSTN/PLMN part of the network?		
SE 53A:	The network supports the "Special arrangement" procedure for		
5= 00A.	the terminating user?		
SE 54:	HOLD is supported in the PSTN/PLMN part of the network?		
SE 55:	CDIV unconditional is supported in the PSTN/PLMN part of the		
	network?		
SE 55A:	CDIV busy is supported in the PSTN/PLMN part of the network?		
SE 55B:	CDIV no reply is supported in the PSTN/PLMN part of the network?		
SE 55C:	CDIV Mobile subscriber not reachable is supported in the PSTN/PLMN part of the network?		
SE 55D:	CDIV call deflection 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 61A:	Call Completion 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?		ļ

7 Test purposes

7.0 Introduction

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

7.1 Testing of SIP protocol requirements

7.1.1 Test purposes for Basic call, Successful

T	
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.
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?

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

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

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

Test sees number			
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, subset.		
	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 a		
	subset of the parameters is present.		
Configuration			
SIP Parameter	INVITE		
	P-Charging-Vector: <mark>icid-value</mark> ; <mark>orig-ioi</mark>		
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		
	(optional).		
	Repeat this test in reverse direction.		

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

Test case number	SS_bcall_007
Test case group	BCALL/successful
Reference	8/ [21]
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE3
Test purpose	P-Early-Media header supported in early dialogue.
	Larry media neader supported in earry dialogue.
	Ensure that an early dialogue is established by sending a 183 Session Progress
	or 180 Ringing from Network B and the P-Early-Media header is present
	authorizes early media.
Configuration	
SIP Parameter	INVITE
	P-Early-Media: supported
	SDP
	183
	P-Early-Media: [any value authorizes early media]
	SDP
	OR
	P-Early-Media: [any value authorizes early media] SDP
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE
CASE A	
	← 183 Session Progress
	·
CASE B	
	← 180 Ringing
-	Apply post test routine
Comments	Establish a communication from network A to Network B
	Check: Is a 183 or 180 send from Network B to establish an early dialogue?
	Check: Is an SDP present in the 183 as a SDP answer?
	Check: Is a bearer transmission possible in backward direction?
	(optional). NOTE: The absence of the direction parameter of an 'a' line represents the
	default value 'sendrecv'.
	NOTE: The presence of the P-Early-Media header in the INVITE request
	indicates the support of "early media Authorization" in the originating
	Network.
	NOTE: The presence of the P-Early-Media header in the 183 or 180 indicates
	the support of the P-Early-Media header and authorizes the media in
	the early dialogue.
	Repeat this test in reverse direction.
	Repeat this test by a call setup to an announcement application.
	in the second of a ball becap to an announcement application.

Test case number	SS bcall 008
Test case group	BCALL/successful
Reference	8/ [21]
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE 3 AND (SE 25 OR SE 26 OR SE 27 OR
	SE 28 OR SE 29)
Test purpose	P-Early-Media header supported early dialogue with 181.
	Ensure that an early dialogue is established by sending a 181 Call Is Being
	Forwarded from Network B and the P-Early-Media header is present authorizes
	early media. The Call is forwarded in network B.
Configuration	Subscription options:
C	 Originating user receives notification that his communication has been
	diverted = <mark>Yes</mark>
SIP Parameter	INVITE
	P-Early-Media: supported
	SDP
	181
	P-Early-Media: [any value authorizes early media]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE -
	 ← 181 Call Is Being Forwarded
	Apply post test routine
Comments	Establish a communication from network A to Network B
	Check: Is a 181 sent from Network B to establish an early dialogue?
	Check: Is an SDP present in the 181 as a SDP answer? Check: Is a bearer transmission possible in backward direction?
	(Optional).
	NOTE: The presence of the P-Early-Media header in the INVITE request
	indicates the support of "early media Authorization" in the originating
	Network.
	NOTE: The presence of the P-Early-Media header in the 181 indicates the
	support of the P-Early-Media header and authorizes the media in the
	early dialogue.
	Repeat this test in reverse direction.

Test case number	SS_bcall_009
Test case group	BCALL/successful
Reference	8/ [21]
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE 3 AND SE 35
Test purpose	P-Early-Media header supported early dialogue with 182.
	Ensure that an early dialogue is established by sending a 182 Queued from
	Network B and the P-Early-Media header is present authorizes early media. The
	Call is a waiting call in network B.
Configuration	
SIP Parameter	INVITE
	P-Early-Media: supported
	SDP
	182
	P-Early-Media: [any value authorizes early media]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE ->
	← 182 Queued
	Apply post test routine
Comments	Establish a communication from network A to Network B
	Check: Is a 181 sent from Network B to establish an early dialogue?
	Check: Is an SDP present in the 182 as a SDP answer? Check: Is a bearer transmission possible in backward direction?
	(Optional). NOTE: The presence of the P-Early-Media header in the INVITE request
	indicates the support of "early media Authorization" in the originating
	Network.
	NOTE: The presence of the P-Early-Media header in the 182 indicates the
	support of the P-Early-Media header and authorizes the media in the
	early dialogue.
	Repeat this test in reverse direction.

Test case number	SS_bcall_010
Test case group	BCALL/successful
Reference	5.10/ [2]
SELECTION EXPRESSION	
Test purpose	Record-route header in the INVITE.
	Ensure that if the Record-Route header is present in the INVITE establishes a communication between a user of network A and a user of network B the topmost header is set to the IBCF of network A.
Configuration	
SIP Parameter	INVITE
	Record-Route: <address a="" ibcf="" in="" network="" of=""></address>
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → Apply post test routine
Comments	 Establish a communication from network A to Network B Check: If present the topmost Record-Route header or entry contains the address of the IBCF of network A. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS bcall 011
Test case group	BCALL/successful
Reference	5.10/ [2]
	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_012
Test case group	BCALL/successful
Reference	5.10/ [2]
SELECTION EXPRESSION	
Test purpose	Record-Route header in the 180 Ringing.
	Ensure if a Record-Route header was present in the initial INVITE that the Record-Route header is present in the 180 Ringing provisional response as the first response from network B upon a connection establish setup from network A.
Configuration	
SIP Parameter	INVITE Record-Route 180: Record-Route
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing Apply post test routine
Comments	Establish a communication from network A to Network B Check: If the Record-Route header is present is in the 180 Ringing. NOTE: The Record-Route header is optional. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_013			
Test case group	BCALL/successful			
Reference	5.10/ [2]			
SELECTION EXPRESSION				
Test purpose	Route header in the BYE of the originating user.			
	Ensure that if a Record-Route header was present in the initial INVITE the Route			
	header may be present in the BYE request sent from the originating user			
	equipment in network A the topmost Route header or entry is set to the IBCF of network B.			
Configuration				
SIP Parameter	BYE:			
	Route: <address b="" ibcf="" in="" network="" of="">;Ir,</address>			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	A confirmed session already exists			
	BYE 🗲			
	← 200 OK BYE			
	Apply post test routine			
Comments	Establish a communication from network A to Network B			
	Check: Is the Route header present in the BYE, the topmost header or entry is set to the address of the IBCF of network B.			
	Repeat this test in reverse direction.			
	Repeat this test with all chosen end devices.			

Test sees would be				
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 terminating user.			
	Ensure that if a Record-Route header was present in the initial INVITE the Route header may be present in the BYE request sent from the terminating user equipment in network B the topmost Route header or entry is set to the IBCF of network A.			
Configuration				
SIP Parameter	BYE:			
	Route: <address a="" ibcf="" in="" network="" of="">;Ir,</address>			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	A confirmed session already exists			
	200 OK BYE →			
	Apply post test routine			
Comments	Establish a communication from network A to Network B			
	Check: If the Route header present in the BYE, the topmost header or entry is			
	set to the address of the IBCF of network A.			
	Repeat this test in reverse direction.			
	Repeat this test with all chosen end devices.			

r					
Test case number	SS_bcall_015				
Test case group	BCALL/successful				
Reference	5.10/ [2]				
SELECTION EXPRESSION					
Test purpose	Route header in the ACK.				
	Ensure that if a Record-Route header was present in the initial INVITE the Route				
	header may be present in ACK from network A upon a connection establishment				
	from network A is completed the topmost Route header or entry is set to the				
	IBCF of network B.				
Configuration					
SIP Parameter	ACK:				
	Route: <address b="" ibcf="" in="" network="" of="">;Ir,</address>				
Message flow					
SIP (Network A)	Interconnection Interface SIP (Network B)				
	INVITE 🗲				
	← 180 Ringing				
	← 200 OK INVITE				
	ACK →				
	Apply post test routine				
Comments	Establish a communication from network A to Network B				
	Check: Is the Route header present in the ACK, the topmost header or entry				
	is set to the address of the IBCF of network B.				
	Repeat this test in reverse direction.				
	Repeat this test with all chosen end devices.				

Test case number	SS bcall 016				
Test case group	BCALL/successful				
Reference	[4] and [5]				
SELECTION EXPRESSION					
Test purpose	Handling of SDP parameters in the INVITE.				
	Ensure that call establishment and the correct handling of the SDP parameters				
	of the INVITE message defined as: TYPE_SDP is performed correctly. Ensure				
	that in the active call state the voice/data transfer on the media channels is				
	performed correctly (e.g. testing QoS parameters). In case when the parameter				
	in the SDP rtpmap: <dynamic-pt> is used the codecs in table 7.1.1-1 applies.</dynamic-pt>				
Configuration					
SIP Parameter	INVITE:				
	Content-Type: application/sdp				
	m=audio <port number=""> RTP/AVP TYPE_SDP= PIXIT (table 7.1.1-1)</port>				
	or				
	m= Image <port number=""> Udptl or Tcptl TYPE_SDP= PIXIT (table 7.1.1-1)</port>				
	a= <mark>TYPE_SDP</mark> = PIXIT (table 7.1.1-1)				
Magazza flaur	b=TYPE_SDP= PIXIT (table 7.1.1-1)				
Message flow	Interconnection Interface CID (Network D)				
SIP (Network A)	Interconnection Interface SIP (Network B)				
	Apply post test routine				
Comments	Establish a communication from network A to Network B				
	Check: Is the preferred codec set to TYPE_SDP?				
	Check: If present: Is the a line set to TYPE_SDP?				
	Check: If present: Is the b line set to TYPE_SDP?				
	Check: Is the codec list consistent with the attribute(s) (bandwidth) regarding				
	the media description?				
	Repeat this test in reverse direction.				
	Repeat this test with all chosen end devices.				

Test case number	SS_bcall_017			
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 an SDP with the offer 1 according table 7.1.1-1.			
	Ensure that answer related to the SDP offer is contained in the 200 OK INVITE			
	message.			
	Ensure that in the confirmed state the voice transfer on the media and B-			
	channels is performed correctly.			
Configuration				
SIP Parameter				
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE (SDP1) →			
	← 180 Ringing			
	← 200 OK INVITE (SDP2)			
	ACK 🗲			
Apply post test routine				
Comments	Establish a communication from network A to Network B.			
	Check: Is the SDP answer contained in the 200 OK INVITE?			
	NOTE: An SDP answer could be present in a provisional response.			
	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.				

TYPE_SDP m= line		b= line		a= line	
VA	<media></media>	<transport></transport>			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>
VA_06	audio	RTP/AVP	Dynamic PT		rtpmap: <dynamic-pt> AMR- WB/16000/1</dynamic-pt>
VA_07	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> AMR/8000/1</dynamic-pt>
NOTE:	<bandwic< td=""><td>lth value> for <n< td=""><td>nodifier> of AS</td><td>is evaluated to be B kbit/s.</td><td></td></n<></td></bandwic<>	lth value> for <n< td=""><td>nodifier> of AS</td><td>is evaluated to be B kbit/s.</td><td></td></n<>	nodifier> of AS	is evaluated to be B kbit/s.	

Table 7.1.1-1

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.		
Quality and the second s	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. The call establishment procedures based on SIP/SDP and H.248 for a real-time fax over IP service are described in Recommendation ITU-T Q.4016 [28].		
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) 		
	← 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 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.				
	rax 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.				
	The call establishment procedures based on SIP/SDP and H.248 for a real-time				
	fax over IP service are described in Recommendation ITU-T Q.4016 [28].				
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=qpmd; vbd=ves				
	a-gpma, vou-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					
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 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.		
	The call establishment procedures based on SIP/SDP and H.248 for a real-time		
	fax over IP service are described in Recommendation ITU-T Q.4016 [28].		
Configuration			
SIP Parameter	INVITE: SDP		
	m=audio <port> RTP/AVP 8 OR <dynamic-pt></dynamic-pt></port>		
	a=rtpmap 8 OR <dynamic-pt> PCMA/8000</dynamic-pt>		
	m=image <port> udptl t38</port>		
	180/200 OK INVITE: SDP		
	m=image <port> udptl t38</port>		
Message flow			
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 successful?		
	Repeat this test in reverse direction.		

Test case number	SS bc	all 023		
Test case group	BCALL/successful			
Reference	4.9 and annex N/ [2]			
SELECTION EXPRESSION		rk A] SE 47 AND [Network A] SE 4 Al	ND [Netwo	ork B] SE 4
Test purpose		p sending, the Multiple INVITE met		
	F			
		that call establishment using overlap that in the confirmed state the voice		
		ormed correctly.		IT the media and B-channels
Configuration				
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(CSq 1)	→	, , , , , , , , , , , , , , , , , , ,
		INVITE(CSq 2)	→	
	+	484 Address Incomplete(CSq 1) ACK	→	
		INVITE(CSq 3)	→	
	←	484 Address Incomplete(CSq 2)		
		ACK	→	
	INVITE(CSq 4) →			
	← 484 Address Incomplete(CSq 3)			
		ACK	→	
	+	180 Ringing(CSq 4)		
		Apply post test routine		
Comments	Check	sh a communication from ISDN to SIF : All INVITE requests contain the s swers with 180 Ringing.		
		t this test in reverse direction.		

Test case number	SS_bcall_024				
Test case group	BCALL/successful				
Reference	4.9 and annex N/ [2]				
SELECTION EXPRESSION	[Network A] SE 47 AND [Network A] SE 5 AND [Network B] SE 5				
Test purpose					
Test huthose	Overlap sending, the in-Dialogue method is used Ensure that call establishment using overlap sending is performed correctly. Ensure that in the confirmed state the voice transfer on the media and B-channels is performed correctly.				
Configuration					
SIP Parameter	INVITE 2: Supported: 100rel				
	183: Require: 100rel				
	INFO: Content-Type: application/x-session-info SubsequentDigit: <additional digits=""></additional>				
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(CSq 1) 1 → 484 Address Incomplete(CSq 1) → ACK → INVITE(CSq 2) 2 → € 183 Session Progress(CSq 2) → PRACK → € 200 OK PRACK → INFO → € 200 OK INFO → INFO → € 200 OK INFO → INFO → INFO → 180 Ringing(CSq 2) →				
Comments	Apply post test routine Establish a communication from ISDN to SIP using the overlap operation in ISDN Check: All INVITE requests contains the same Call ID and From header values. Check: The 183 session Progress that establishes an early dialogue contains a Require header set to 100rel. Check: All INFO requests contain the Content-Type header set to 'application/x-session-info'. Check: All INFO requests contains the 'SubsequentDigit.' MIME body containing the additional digits. The UE B answers with 180 Ringing response after the INVITE was received. Repeat this test in reverse direction.				

Test sees would be	00 1		
Test case number	SS_bcall		
Test case group		successful	
Reference	5.1.1.1.2		
SELECTION EXPRESSION		A] (SE 46 OR SE 47) AND [Network A] SE 6	
Test purpose	PSTN XML BearerCapability element in the INVITE		
	INVITE r	s located in network A and an ISDN end device is used. Ensure that the request contains a PSTN XML MIME body and a BearerCapability as indicated in table 7.1.1-2 is present.	
Configuration	User A is	s an ISDN access either in the PSTN or the SIP - ISDN interworking g [10] applies	
SIP Parameter	INVITE:		
	Conte	ent-Type: application/vnd.etsi.pstn+xml	
	Content-Disposition: signal;handling=optional		
	xml version="1.0" encoding="utf-8"? PSTN		
	BearerCapability		
	BCoctet3		
	CodingStandard>00<		
	InformationTransferCabability>ITC_value<		
	< BCoctet4		
	TransferMode>00<		
	InformationTransferRate>10000<		
	BCoctet5		
	Layer1Identification>01<		
	UserInfoLayer1Protocol>00011<		
Message flow			
SIP (Network A)		Interconnection Interface SIP (Network B)	
		Apply post test routine	
Comments	Check:	Is a PSTN XML MIME body contained in the INVITE request?	
	Check:		
	Check:	Is the InformationTransferCabability element set as indicated in	
		table 7.1.1-2?	
	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 t	this test in reverse direction.	

Table 7.1.1-2: PSTN XML BearerCapability

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

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

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

Test case number	SS_bcall_028		
Test case group	BCALL/successful		
Reference	5.1.2.2/ [25]		
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6		
Test purpose	PSTN XML ProgressIndicator element in the 180.		
	· · · · · · · · · · · · · · · · · · ·		
	User B is located in network B and an ISDN end device is used. Ensure that the		
	180 Ringing response contains a PSTN XML MIME body and at least one		
	ProgressIndicator element is present.		
Configuration	User B is an ISDN access either in the PSTN or the SIP - ISDN interworking		
	according to [10] applies		
SIP Parameter	180:		
	Content-Type: application/vnd.etsi.pstn+xml		
	Content-Disposition: signal;handling=optional		
	-2vml version_"1.0" encoding_"utf 8"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)		
	$\leftarrow 180 \text{ Ringing}$		
	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 '0000111'?		
	Check: Is optional a second ProgressIndicator element present and the		
	ProgressDescription element is set to any value not #2 and not #8?		
	Repeat this test in reverse direction.		

Test case number	SS_bcall_029		
Test case group	BCALL/successful		
Reference	5.1.2.3/ [25]		
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6		
Test purpose	PSTN XML ProgressIndicator element in the 200.		
	User B is located in network B and an ISDN end device is used. Ensure that the		
	200 OK INVITE response contains a PSTN XML MIME body and at least one		
	ProgressIndicator element is present.		
Configuration	User B is an ISDN access either in the PSTN or the SIP - ISDN interworking		
	according to [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 '0000111'?		
	Repeat this test in reverse direction.		

Test case number	SS_bcall	_030	
Test case group	BCALL/s		
Reference	5.1.1.1.2/	/ [25]	
SELECTION EXPRESSION	[Network	A] (SE 46 OR SE 47) AND [Network A] SE 6	
Test purpose	PSTN XML BearerCapability Fallback connection type element in the		
	INVITE.		
		located in network A and an ISDN end device is used. Ensure that the	
		equest contains a PSTN XML MIME body and one BearerCapability	
		s present the InformationTransferCabability element is set to '00000'	
		nformationTransferCabability element is set to '10001'.	
Configuration	User A is an ISDN access either in the PSTN or the SIP - ISDN interworking		
		g to [10] applies	
SIP Parameter	INVITE:		
		ent-Type: application/vnd.etsi.pstn+xml	
	Content-Disposition: signal;handling=optional		
	xml version="1.0" encoding="utf-8"? PSTN		
		arCanability	
	BearerCapability BCoctet3		
	CodingStandard>00<		
	InformationTransferCabability>00000<		
	BearerCapability		
	BCoctet3		
	CodingStandard>00<		
		InformationTransferCabability>10001<	
Message flow			
SIP (Network A)		Interconnection Interface SIP (Network B)	
		INVITE >	
Apply post test routine			
Comments	Check:	Is a PSTN XML MIME body contained in the INVITE request?	
	Check:	Is the first BearerCapability InformationTransferCabability element set	
	0	as indicated to '00000'?	
	Check:	Is the second BearerCapability InformationTransferCabability element	
	Check:	set as indicated to '10001'? Is the InformationTransferCabability element value consistent with the	
	CHECK:	codec list in the SDP?	
	Check:	Is the InformationTransferCabability element value consistent with the	
	SHOOK.	bandwidth information in the SDP?	
	Repeat th	his test in reverse direction.	

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

Test case number	SS_bcall	032	
Test case group	BCALL/successful		
Reference	5.1.2.3/ [25]		
SELECTION EXPRESSION	-	B] (SE 46 OR SE 47) AND [Network B] SE 6	
Test purpose		<pre>c coccurs.</pre>	
		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	
		TE response contains a PSTN XML MIME body and a BearerCapability is present the InformationTransferCabability element set to '00000'. A	
		/L MIME ProgressIndicator body is present, the ProgressDescription is	
	set to '00		
Configuration		an ISDN access either in the PSTN or the SIP - ISDN interworking	
g		g to [10] applies	
SIP Parameter	200:		
		ent-Type: application/vnd.etsi.pstn+xml	
	Conte	ent-Disposition: signal;handling=optional	
	0		
	PSTN	rsion="1.0" encoding="utf-8"?>	
	-	erCapability	
		Coctet3	
	CodingStandard>00<		
		InformationTransferCabability>00000<	
		essIndicator	
	ProgressOctet4		
	ProgressDescription>0000101<		
	OR		
	UK		
	No PSTN	I XML element	
Message flow			
SIP (Network A)		Interconnection Interface SIP (Network B)	
		INVITE ->	
		180 Ringing	
	•	E 200 OK INVITE ACK →	
		Apply post test routine	
Comments	Check:	Is a PSTN XML MIME body contained in the 200 OK INVITE	
		response? OR	
		If no PSTN XML MIME body contained in the 200 OK INVITE	
		response? No further checks!	
	Check:	Is a BearerCapability element present, the	
		InformationTransferCabability element set to '00000'?	
	Check:	Is a ProgressIndicator element present, the ProgressDescription is set	
	Check	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	his test in reverse direction.	

Test case number	SS_bcall_032A		
Test case group	BCALL/successful		
Reference	5.1.2.3/ [26]		
SELECTION EXPRESSION			
Test purpose	Telephony events transmission		
	Ensure that the ability of transmission of Telephony events can be performed by the originating user und the terminating user. The Telephony transmission can be done:		
	 Either by indicating in the SDP offer in the RTP stream Or by the SIP INFO/NOTIFY Method for DTMF tone generation The telephony event transmission applies from the calling user and from the called user as well. 		
Configuration			
SIP Parameter	INVITE: CASE A m=audio <port> RTP/AVP <payload type=""></payload></port>		
	NOTIFY CASE B m=audio <port> RTP/AVP <dynamic-pt> a=rtpmap <dynamic-pt> telephone-event/8000 a=rtpmap <dynamic-pt> 0-15</dynamic-pt></dynamic-pt></dynamic-pt></port>		
	CASE C INFO 2: Content-Type: application/dtmf 'x' or		
	Content-Type: application/dtmf-relay Signal=x Duration=y		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing ← 200 OK INVITE ACK →		
CASE A	RTP DTMF events		
CASE B	INFO 1 → € 200 OK INFO		
CASE C	INFO 2 → € 200 OK INFO		
	Apply post test routine		
Comments	Establish a communication from network A to Network B Check: Case A: Is the dynamic payload type 'telephone-event' present in the		
	SDP offer? Check: Case A: Is the dynamic payload type 'telephone-event' covered in the RTP stream if the Telephone event occurs?		
	 Check: Case B: Does the Content-Type header field in the INFO request conveying the DTMF signal set to 'application/dtmf'? Check: Case B: does the MIME body of the INFO request covering the DTMF 		
	signal contain the events regarding the used content type? Check: Case C : Does the Content-Type header field in the INFO request		
	 conveying the DTMF signal set to 'application/dtmf-relay'? Check: Case C: Does the MIME body of the INFO request covering the DTMF signal contain the events and duration regarding the used content type? 		
	type? Repeat this test in reverse direction.		

Test case number	SS_bcall_032B		
Test case group	BCALL/successful		
Reference	[20]		
SELECTION EXPRESSION	[Network B] SE 17b		
Test purpose	Handling of multiple early dialogues.		
	······································		
	Ensure that in case of forking in Network B the early dialogues are handled in a		
	proper way. When a 200 OK INVITE is received, the remaining early dialogues		
	shall be cancelled.		
Configuration	User B has registered three end devices under the same identity.		
SIP Parameter			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE >		
	← 180 Ringing 1		
	← 180 Ringing 2		
	← 180 Ringing 3		
	← 200 OK INVITE 3		
	ACK ->		
	Communication		
CASE A			
	← BYE 3		
	200 OK BYE 3 →		
CASE B			
	BYE 3 →		
	← 200 OK BYE 3		
Comments	Establish a communication from network A to Network B Check: Ensure that several provisional responses with different 'To' tags are sent from Network B to Network A.		
L	Repeat this test in reverse direction.		

Test case number	SS_bcall_033	
Test case group	BCALL/successful	
Reference	7.1/[24]	
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] SE 47	
Test purpose	SIP-I support, Basic call, IAM present in the INVITE request.	
	Ensure that when a call initiated in the PSTN or the PLMN and the ISUP - SIP-I interworking is applicable in the originating network, an ISUP IAM is encapsulated in the initial INVITE request.	
	Ensure that all the mandatory parameters in the IAM are present and the values are valid and the Transmission medium requirement parameter is consistent	
	with the SDP.	
Configuration		
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name]	
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required	
Maaaaaa faay	IAM Nature of connection indicators Forward call indicators Calling party's category Transmission medium requirement Called party number Calling party number (optional) Optional forward call indicators (optional) Hop counter (optional) User service information (optional) Access transport (optional) [any boundary name]	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(IAM) → ← 100 Trying	
Commonto	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 with all possible IAM USI and ATP combinations as indicated in table 7.1.1-3	
	Repeat this test in reverse direction.	

Table 7.1.1-3: IAM Parametrization

ITC_value	IAM USI	ATP
ITC_VA_1	Speech	HLC: telephony
ITC_VA_2	3,1 kHz audio	No HLC
ITC_VA_3	3,1 kHz audio	HLC: facsimile group 2/3
ITC_VA_4	3,1 kHz audio	LLC: 3,1 kHz audio, voice band data via modem,
		synchronous mode, user rate 2,4 kbit/s
ITC_VA_5	unrestricted digital information	HLC: facsimile group 4
ITC_VA_6	unrestricted digital information	HLC: facsimile group 4, LLC: telematic_term
ITC_VA_7	Speech	No HLC
ITC_VA_8	unrestricted digital information	No HLC

Test case number	SS bcall	034	
		_	
Test case group		uccessful	
Reference	7.2.1/[24		
SELECTION EXPRESSION		A] SE 4 AND SE 17 AND [Net	
Test purpose	SIP-I support, Basic call, overlap signalling.		nalling.
	Ensure th	nat when overlap signalling app	lies in the ISUP -SIP-I interworking in the
	originatin	g network, several INVITE req	uests with the same Cal-ID and From tag
	are sent	from Network A to Network B.	
	Ensure th	nat the original IAM is encapsu	ated in any INVITE request.
Configuration			
SIP Parameter			
Message flow	•		
SIP (Network A)		Interconnection Interface INVITE(1)	SIP (Network B) ➔
	←	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 networ	k A to Network B using the overlap
	procedur	e in Network A	
	Check:		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 rea	
	Check:		ent in the initial INVITE request also
		encapsulated in any following	INVITE request required for the call
		setup?	
	Repeat the	his 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 [Network B] SE 47	
Test purpose	SIP-I support, Basic call, ACM present in the 180 response.	
rest purpose	SIF-I Support, Basic call, ACM present in the roo response.	
	Ensure that on receipt of a 180 Ringing provisional response and an	
	SIP-I - ISUP interworking is applicable in the terminating network the Backward	
	call indicators parameter in the encapsulated ACM is present and the values are	
	valid.	
	Ensure that the values of the optional parameters in the encapsulated ACM are	
	valid.	
Configuration	Valia.	
SIP Parameter	180:	
SIF Farameter	Content-Type: multipart/mixed;boundary=[any boundary name]	
	Contenter ype. Inditipatornixed, boundary-lany boundary namej	
	[any boundary name]	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	ACM	
	Backward call indicators	
	Called party's status indicator= subscriber free	
	[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?	
	Check: Can the ringing tone be heard from the terminating side?	
1	Repeat this test in reverse direction.	
	הפורפת נוווש נכשו וו ופעפושם מוופטנוטוו.	

Test case number	SS_bcall_036	
Test case group	BCALL/successful	
Reference	6.5/ [24]	
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] 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 Optional backward call indicator Inband info or appropriate pattern is now available Access Transport (optional) Progress Indicator Progress description = Destination address is non ISDN	
	[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: Can an early media (e.g. announcement) be heard from the 	
	terminating side? 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 [Network B] 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	
Configuration	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 houndary name]	
	[any boundary name]	
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required	
	Contont Disposition. signal, nanoning-required	
	CPG	
	Event indicator = ALERTING	
	[any boundary name]	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
	INVITE +	
	← 100 Trying	
	← 183 Session Progress(ACM)	
	 180 Ringing(CPG) 	
	Apply post test routine	
Comments	Establish a communication from network A to Network B	
	Check: Is an ISUP CPG message encapsulated in the 180 Ringing provisional	
	response?	
	Check: Is the mandatory Event indicator present in the encapsulated ISUP CPG set to 'ALERTING'?	
	Check: Are the values of optional parameters in the encapsulated ISUP CPG valid?	
	Repeat this test in reverse direction.	

Test case number	SS_bcall_038
Test case group	BCALL/successful
Reference	6.7/ [24]
SELECTION EXPRESSION	[Network B] SE 17 [Network B] 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	200 OK: Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name]
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ANM
	[any boundary name]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE →
	 ← 100 Trying ← 180 Ringing(ACM)
	 ← 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.
	וויפיפוג נווט נפט ווו ופיפוטב עוופטוטוו.

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 [Network A] AND SE 47		
Test purpose	SIP-I support. Basic call, REL present in a BYE request sent from the originating network.		
	Ensure that an ISUP REL message is encapsulated in a BYE request sent in the release procedure initiated from the originating user when ISUP - SIP-I interworking is applicable in the originating network. Ensure the validity of the cause indicator in the encapsulated REL. Ensure that the ISUP RLC is encapsulated in the 200 OK BYE.		
Configuration			
SIP Parameter	BYE: Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name]		
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	REL Cause value:		
	[any boundary name]		
	200 OK BYE Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	RLC		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → € 100 Trying € 180 Ringing € 200 OK INVITE ACK → Communication BYE(REL) → € 200 OK BYE(RLC)		
Comments	 Establish a confirmed communication from network A to Network B The 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 [Network B] AND SE 47		
Test purpose	SIP-I support. Basic call, REL present in a BYE request sent from the		
	terminating network.		
	Ensure that an 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		
	Content-Disposition. signal, nationing=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		
	Content-Disposition. signal, nanuling=required		
	RLC		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE ->		
	← 100 Trying		
	← 180 Ringing		
	← 200 OK INVITE		
	ACK ->		
	Communication		
Commonto	$200 \text{ OK BYE(RLC)} \rightarrow$		
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: Is the ISOP REL encapsulated in the BYE request? Check: Are the cause indicators in the encapsulated ISUP REL valid?		
	Check: If a Reason header is present in the BYE request, is the 'cause' value		
	of Reason header equal to the 'Cause value' in the encapsulated		
	REL?		
	Check: Is the ISUP RLC encapsulated in the 200 OK BYE?		
	Repeat this test in reverse direction.		

7.1.2 Codec negotiation

Reference [3], [4] and SELECTION EXPRESSION Test purpose Session u	bdec_Negotiation		
Reference [3], [4] and SELECTION EXPRESSION Test purpose Session u			
SELECTION EXPRESSION Test purpose Session u	. [+]		
Test purpose Session u			
During the	Session update requested by the calling user.		
media ses containing existing di session in to accept In case wh	e session, the calling user decides to change the characteristics of the sion. This is accomplished by sending a re-INVITE or UPDATE or UPDATE a new media description. This re-INVITE or UPDATE references the alogue so that the other party knows that it has to modify an existing stead of establishing a new session. The other party sends a 200 (OK) the change. The requestor responds to the 200 (OK) with an ACK. The parameter in the SDP rtpmap: <dynamic-pt> is used the table 7.1.2-1 applies.</dynamic-pt>		
Configuration			
SIP Parameter SDP1: cod	SDP1: codec x chosen from table 7.1.2-1 SDP3: codec y chosen from table 7.1.2-1		
	Interconnection Interface SIP (Network B) ned session already exists (SDP 1)		
CASE A	INVITE(SDP3) → 200 OK INVITE(SDP4) ACK		
CASE B	UPDATE(SDP3) → 200 OK UPDATE(SDP4)		
	Apply post test routine		
from the ta Check: Check:	a communication from network A to Network B using SDP1 chosen able 7.1.2-1 The calling user changes the media description using INVITE request containing SDP 3 codec chosen from table 7.1.2-1 different to SDP1. Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description?		
	is test in reverse direction.		

48

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 dialogue so that the other party knows that it has 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) A confirmed session already exists (SDP 1)		
CASE A	INVITE(SDP3) → CONTE(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 an SDP with the offer 1. 		
	 Ensure that answer related to the SDP offer is contained in the 200 OK INVITE message. 		
	Ensure that in the confirmed call state the voice transfer on the media channels		
O and income the m	is performed correctly.		
Configuration			
SIP Parameter	INVITE: SDP offer		
	200: SDP answer		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(SDP1) →		
	← 180 Ringing ← 200 OK INVITE(SDP2)		
	← 200 OK INVITE(SDP2) ACK →		
	Apply post test routine		
Comments	Establish a communication from network A to Network B		
	Check: Is the SDP offer contained in the initial INVITE request?		
	Check: Is the SDP answer contained in the 200 OK INVITE final response?		
	NOTE: An SDP answer could be present in a provisional response.		
	Repeat this test in reverse direction.		

VARIABLE	PT	Encoding	Media	Clock	Channels	Supported in	Supported in
			type	rate		network A	network B
VA_01	0	PCMU	А	8 000	1		
VA_02	3	GSM	А	8 000	1		
VA_03	4	G723	А	8 000	1		
VA_04	5	DVI4	А	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	А	44 100	2		
VA_10	11	L16	А	44 100	1		
VA_13	12	QCELP	А	8 000	1		
VA_12	13	CN	А	8 000	1		
VA_13	14	MPA	А	90 000			
VA_14	15	G728	А	18 000	1		
VA_15	16	DVI4	A	11 025	1		
VA_16	17	DVI4	А	22 050	1		
VA_17	18	G729	А	8 000	1		
VA_18	Dyn	G726-40	А	8 000	1		
VA_19	Dyn	G726-32	A	8 000	1		
VA_20	Dyn	G726-24	A	8 000	1		
VA_21	Dyn	G726-16	A	8 000	1		
VA_22	Dyn	G729D	Α	8 000	1		
VA_23	Dyn	G729E	А	8 000	1		
VA_24	Dyn	GSM-EFR	А	8 000	1		
VA_25	25	CelB	V	90 000			
VA_26	26	JPEG	V	90 000			
VA_27	28	Nv	V	90 000			
VA_28	31	H261	V	90 000			
VA_29	32	MPV	V	90 000			
VA_30	33	MP2T	V	90 000			
VA_31	34	H263	V	90 000			
VA_32	Dyn	H263-1998	V	90 000			
VA_33	Dyn	AMR	А	8 000	1		
VA_34	Dyn	AMR-WB	А	16 000	1		
VA_35	Dyn	telephone- event	A	8 000	1		

Table 7.1.2-1

7.1.3 Resource Reservation

Test sees number			
Test case number	SS_resource_001		
Test case group	BCALL/Resource_Reservation		
Reference	[3], [4], [5] and [6]		
SELECTION EXPRESSION	([Network A] SE 7 AND [Network B] SE 7) AND ([User A] SE 42 AND [User B]		
	SE 42)		
Test purpose	Resource reservation successful, segmented status.		
	 Ensure that the network is able to reserve resources for quality of service when requested from the initiating user: In the INVITE 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			
een galaaten			

SIP Parameter	INVITE: Supported: 100rel precondition
	SDP1: m=audio <port number=""> RTP/AVP <codec></codec></port>
	a=curr:qos local none
	a=curr:qos remote none
	a=des:qos mandatory/optional local sendrecv
	a=des:qos none remote sendrecv
	183 Session Progress: Supported/Require: 100rel precondition
	SDP2: m=audio <port number=""> RTP/AVP <codec></codec></port>
	a=curr:gos local none
	a=curr:qos remote none
	a=des:gos mandatory/optional local sendrecv
	a=des:qos mandatory/optional remote sendrecv
	UPDATE
	SDP3: m=audio <port number=""> RTP/AVP <codec></codec></port>
	a=curr:qos local sendrecv
	a=curr:qos remote none
	a=des:qos mandatory/optional local sendrecv
	a=des:gos mandatory/optional remote sendrecv
	200 OK UPDATE
	SDP4: a=curr:gos local sendrecv
	a=curr:qos remote sendrecv
	a=des:qos mandatory/optional local sendrecv
	a=des:qos mandatory/optional 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) →
Comments	← 200 OK UPDATE(SDP4)
Commenta	Apply post test routine
	Apply post test routine Establish a communication from network A to Network B
	Apply post test routine Establish a communication from network A to Network B Check: Is the quality of service for the current state local and remote set to
	Apply post test routine 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?
	Apply post test routine 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
	Apply post test routine Establish a communication from network A to Network B Check: Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the INVITE? Check: Is the quality of service for the desired state local and remote set to 'mandatory' and 'sendrecv' in the 183?
	Apply post test routine Establish a communication from network A to Network B Check: Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the INVITE? Check: Is the quality of service for the desired state local and remote set to 'mandatory' and 'sendrecv' in the 183? Check: Is the quality of service for the current state local set to 'sendrecv'
	Apply post test routine Establish a communication from network A to Network B Check: Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the INVITE? Check: Is the quality of service for the desired state local and remote set to 'mandatory' and 'sendrecv' in the 183? Check: Is the quality of service for the current state local set to 'sendrecv' indicated in the SDP in the UPDATE?
	Apply post test routine Establish a communication from network A to Network B Check: Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the INVITE? Check: Is the quality of service for the desired state local and remote set to 'mandatory' and 'sendrecv' in the 183? Check: Is the quality of service for the current state local set to 'sendrecv' indicated in the SDP in the UPDATE? Check: Is the quality of service for the current state local and remote set to
	Apply post test routine Establish a communication from network A to Network B Check: Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the INVITE? Check: Is the quality of service for the desired state local and remote set to 'mandatory' and 'sendrecv' in the 183? Check: Is the quality of service for the current state local set to 'sendrecv' indicated in the SDP in the UPDATE? Check: Is the quality of service for the current state local and remote set to 'sendrecv' indicated in the SDP in the 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 SDP in the 200 OK UPDATE?
	Apply post test routine Establish a communication from network A to Network B Check: Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the INVITE? Check: Is the quality of service for the desired state local and remote set to 'mandatory' and 'sendrecv' in the 183? Check: Is the quality of service for the current state local set to 'sendrecv' indicated in the SDP in the UPDATE? Check: Is the quality of service for the current state local and remote set to 'sendrecv' indicated in the SDP in the SDP in the 200 OK UPDATE? Check: Is the codec in the codec list consistent with the attribute(s)
	Apply post test routine Establish a communication from network A to Network B Check: Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the INVITE? Check: Is the quality of service for the desired state local and remote set to 'mandatory' and 'sendrecv' in the 183? Check: Is the quality of service for the current state local set to 'sendrecv' indicated in the SDP in the UPDATE? Check: Is the quality of service for the current state local and remote set to 'sendrecv' indicated in the SDP in the 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? Check: Is the codec in the codec list consistent with the attribute(s) (bandwidth) regarding the media description? At least a G.711 codec
	Apply post test routine Establish a communication from network A to Network B Check: Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the INVITE? Check: Is the quality of service for the desired state local and remote set to 'mandatory' and 'sendrecv' in the 183? Check: Is the quality of service for the current state local set to 'sendrecv' indicated in the SDP in the UPDATE? Check: Is the quality of service for the current state local and remote set to 'sendrecv' indicated in the SDP in the 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? Check: Is the codec in the codec list consistent with the attribute(s) (bandwidth) regarding the media description? At least a G.711 codec is required.
	Apply post test routine Establish a communication from network A to Network B Check: Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the INVITE? Check: Is the quality of service for the desired state local and remote set to 'mandatory' and 'sendrecv' in the 183? Check: Is the quality of service for the current state local set to 'sendrecv' indicated in the SDP in the UPDATE? Check: Is the quality of service for the current state local and remote set to 'sendrecv' indicated in the SDP in the 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? Check: Is the codec in the codec list consistent with the attribute(s) (bandwidth) regarding the media description? At least a G.711 codec

Test case number	SS_resource_002			
Test case group	BCALL/Resource_Reservation			
Reference	[3], [4], [5] and [6]			
SELECTION EXPRESSION	(Network A] SE 7 AND ([User A] SE 42 AND NOT [User B] SE 42)			
Test purpose	Resource reservation not supported.			
	 Ensure that the network is able to reserve resources for quality of service when requested from the initiating user. The terminating user dies not support the precondition procedure. In the INVITE the UE requests to establish QoS preconditions for all the media streams. In the 183 Session Progress: no support by the terminating UA is indicated. Or In the 180 Ringing: no support by the terminating UA is indicated. Or 			
	In the 200 OK INVITE: no support by the terminating UA is indicated.			
Configuration				
SIP Parameter	INVITE: Supported: 100rel precondition SDP1: m=audio <port number=""> RTP/AVP <codec> a=curr:qos local none a=curr:qos remote none a=des:qos mandatory/optional local sendrecv a=des:qos none remote sendrecv</codec></port>			
	183 Session Progress: SDP2: m=audio <port number=""> RTP/AVP <codec> Or 180 Ringing: SDP2: m=audio <port number=""> RTP/AVP <codec> Or 200 OK: SDP2: m=audio <port number=""> RTP/AVP <codec></codec></port></codec></port></codec></port>			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE(SDP1)			
CASE A	← 183 Session Progress(SDP2)			
CASE B	← 180 Ringing(SDP2)			
CASE C	 ← 180 Ringing ← 200 OK INVITE(SDP2) ACK → 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 support of Precondition not indicated in the 183 Session Progress (optional) Check: Is the support of Precondition not indicated in the 180 Ringing 			
	(optional) Check: Is the support of Precondition not indicated in the 200 OK INVITE			
	NOTE: This test case is applicable with an VoLTE originator.			

SS_unsucc_001			
BCALL/unsuccessful			
[4]			
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.			
Interconnection Interface SIP (Network B)			
INVITE ->			
← 404 Not Found			
ACK →			
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?			
Check: In case of an interworking into the PSTN, is a Reason header			
cause 1 present in the 404 Not Found?			
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 is 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				
Comments	ACK → Establish a communication from network A to Network B, Network B is unable to			
Comments	ACK → Establish a communication from network A to Network B, Network B is unable to process the request.			
Comments	ACK → 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? Check: In case of an interworking into the PSTN, is a Reason header			

Teet eeee number	
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?
	Check: In case of an interworking into the PSTN, is a Reason header
	cause 17 present in the 486 Busy Here?
	Repeat this test in reverse direction.
	Repeating test in reverse direction

Tast sass much su	00		
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? Check: In case of an interworking into the PSTN, is a Reason header cause 17 present in the 486 Busy Here? 		
	Repeat this test in reverse direction.		

Test case number	SS_unsucc_005		
Test case group	BCALL/unsuccessful		
Reference	[4]		
SELECTION EXPRESSION			
Test purpose	The called user is not available on the called number.		
	Ensure that when the number is changed, the network initiate call clearing		
	to the calling user with a 410 Gone message.		
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE ->		
	← 410 Gone		
	ACK →		
Comments	Establish a communication from network A to Network B, user B is not		
	allocated in Network B.		
	Check: Is a 410 Gone sent from Network B to Network A?		
	Check: In case of an interworking into the PSTN, is a Reason header		
	present in the 486 Busy Here?		
	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? Check: In case of an interworking into the PSTN, is a Reason header cause 28 present in the 484 Address Incomplete?
1	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 dialogue so that the other party knows that it has to modify an existing session instead of establishing a new session. Ensure that if the other party does not accept the change, it sends an error response such as 488 Not Acceptable Here, which also receives an ACK. The session remains unchanged.	
Configuration		
SIP Parameter	INVITE: codec not supported in Network B	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → € 180 Ringing € 200 OK INVITE ACK → Communication	
	INVITE → 488 Not Acceptable Here ACK → Apply post test routine	
Comments	Establish a communication from network A to Network B. User A in Network A attempts to change the session by sending a SDP offer to the UE in Network B. Network B does not support the codec sent in the offer. Check: Is a 488 Not Acceptable Here sent from Network B to Network A? Repeat this test in reverse direction.	

Test case number	SS_unsucc_008	
Test case group	BCALL/unsuccessful	
Reference	[3], [4] and [5]	
SELECTION EXPRESSION		
Test purpose	Session update requested by the called user is unsuccessful, existing session remains unchanged.	
	During the session, the called user decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialogue so that the other party knows that it has to modify an existing session instead of establishing a new session. Ensure that if the other party does not accept the change, it sends an error response such as 488 Not Acceptable Here, which also receives an ACK. The session remains unchanged. The 488 Not Acceptable Here may be sent by a simulation equipment.	
Configuration		
SIP Parameter	INVITE: codec not supported in Network A	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)	
	INVITE → ← 180 Ringing	
	← 200 OK INVITE	
	ACK → Communication	
	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.	

SS_unsucc_009
BCALL/unsuccessful
[4]
Call clearing due to no answer from the called user initiated by the calling user.
Ensure that when there is no answer from the called user, the calling user initiates call clearing to the called user with CANCEL or BYE
Interconnection Interface SIP (Network B) INVITE →
← 180 Ringing CANCEL/BYE →
 ← 200 OK CANCEL/BYE ← 487 Request Terminated ACK →
 Check: Is a CANCEL or BYE request is sent by the originating user? Check: In case of an interworking into the PSTN, is a Reason header cause 16 present in the CANCEL request? Check: Is a 487 Request Terminating sent by the terminating user? Check: Are the media streams terminated after the 200 OK CANCEL/BYE was sent? Repeat this test in reverse direction.

Test sees would be	00	040		
Test case number	SS_unsucc			
Test case group	BCALL/uns			
Reference	[3], [4] and	[5]		
SELECTION EXPRESSION				
Test purpose	Codec not	supported by the called u	user.	
	called user Ensure tha user initiate	t, when the called user does	s not aco g user wi	s that are not supported by the cept the Media session, the called th 488 Not Acceptable Here or CK.
Configuration		•		
SIP Parameter	INVITE: co	dec not supported at user (Network	B)
Message flow SIP (Network A)	I	Interconnection Interface	→	SIP (Network B)
CASE A	÷	488 Not Acceptable Here ACK	→	
CASE B	+	606 Not Acceptable ACK	→	
Comments	User B in N from Netwo Check: I	call setup from network A t letwork B does not support	the code	ec offered in the SDP received

Test case number	SS_unsucc_011	
_		
Test case group	BCALL/unsuccessful	
Reference	[4]	
SELECTION EXPRESSION		
Test purpose	Call clearing due to no answer from the called user initiated by the originating network.	
	Ensure that when there is no answer from the called user, the originating network initiates 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 → CANCEL/BYE 487 Request Terminated ACK →	
Comments	Check: Is a CANCEL or BYE request sent by the originating network? Check: In case of an interworking into the PSTN, is a Reason header cause 16 present in the CANCEL request?	
	Check: Is a 487 Request Terminating send from the terminating user?Check: Are the media streams terminated after the 200 OK CANCEL/BYE was sent?	
	Repeat this test in reverse direction.	

Test case number	SS_unsucc_011A		
Test case group	BCALL/unsuccessful		
Reference	[27]		
SELECTION EXPRESSION	[Network A) SE 17a AND [Network B] SE 17a		
Test purpose	Negotiation of session timer.		
	Ensure that the interconnected networks are able to negotiate the session time to refresh the session. If the session refresh duration is too short for one of the involved entities, a 422 Session Interval Too Small unsuccessful final response is sent in backward direction to update the session duration time. A new INVITE is sent and a Min-SE header present proposes a longer session duration.		
Configuration	The session time in Network B is smaller than the session time used in Network A		
Comment	This test case is only applicable if the session refresh time is different in Network A and Network B. This situation is also load dependent.		
SIP Parameter	INVITE 1: Supported: timer Session-Expires: x 422: Min-SE. x + y INVITE 2		
	Session-Expires: x + y		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE 1 → 422 Session Interval Too Small ACK		
	INVITE 2 → ← 180 Ringing Apply post test routine		
Comments	 Establish a communication setup from Network A to Network B Check: Is the supported header in the initial INVITE set to 'timer' Check: Is a 422 Session Interval Too Small sent by the terminating Network? Check: Is the Session-Expires header in the second initial INVITE request sent from Network A set to the value indicated in the 422 final response? Repeat this test in reverse direction. 		

Test case number	SS_unsucc_012		
Test case group	BCALL/unsuccessful		
Reference	6.11.2/ [24]		
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47		
Test purpose	SIP-I support. Called number is not allocated in the PSTN/PLMN network.		
	Ensure that, when calling to an unallocated number in the PSTN/PLMN part of		
	network B and ISUP - SIP-I interworking applies in Network B, the network		
	initiate call clearing to the calling user with a 404 Not Found message. An ISUP		
	REL message is encapsulated and the Cause value indicator is set to '1'.		
Configuration	The called user number is not assigned to the PSTN/PLMN part in Network B		
SIP Parameter	404:		
	Reason: Q.850;cause=1 (optional)		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	REL		
	Cause value: 1		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE >		
	← 404 Not Found(REL)		
	ACK →		
Comments	Establish a communication from network A to Network B, called user number is		
	not allocated in the PSTN/PLMN part of Network B		
	Check: Is a 404 Not Found sent from Network B to Network A?		
	Check: Is an ISUP REL encapsulated and the Cause value indicator is set to		
	'1'?		
	Check: If a Reason header is present, is the cause value equal to the value in the Cause value of the appropriated ISUB PEL2		
	the Cause value of the encapsulated ISUP REL?		
	Repeat this test in reverse direction.		

Test case number	SS_unsucc_013	
Test case group	BCALL/unsuccessful	
Reference	6.11.2/ [24]	
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] 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. An 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	
	Cause value. 17	
	[any boundary name]	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
	← 486 Busy Here(REL) ACK →	
Comments	Establish a communication from network A to Network B, user B in the	
Comments	PSTN/PLMN part of Network B is busy.	
	Check: Is a 486 Busy Here sent from Network B to Network A?	
	Check: Is an 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 [Network B] 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. An 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 →	
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 an 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 [Network B] SE 47		
Test purpose	SIP-I support. Call clearing due to no answer from the called user initiated by the calling user.		
	Ensure when the early dialogue is not confirmed by the called user, the calling user located in the PSTN/PLMN part of Network A and ISUP - SIP-I interworking applies in Network A initiates call clearing to the called user with CANCEL or BYE. An ISUP REL message is encapsulated in the BYE request and the Cause value indicator is set to '16'.		
Configuration			
SIP Parameter	480: Reason: Q.850;cause=16 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name]		
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 16		
	[any boundary name]		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →		
CASE A	CANCEL → CANCEL → CANCEL 487 Request Terminated ACK →		
CASE B	BYE(REL) →		
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 an 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: An ISUP REL is not encapsulated in a CANCEL request. 		

Test case number	SS_unsucc_016		
Test case group	BCALL/unsuccessful		
Reference	7.7.1/[24]		
SELECTION EXPRESSION	[Network A] SE 17 AND [Network B] SE 47		
Test purpose	SIP-I support. Call clearing due to no answer from the called user initiated by the originating network.		
	Ensure when the early dialogue is not confirmed by the called user, the originating network initiate the call clearing after timeout of ISUP timer T9 if the calling user is located in the PSTN/PLMN part of Network A and ISUP - SIP-I interworking applies in Network A and the originating network sends a CANCEL or BYE to the called user. An ISUP REL message is encapsulated in the BYE request and the Cause value indicator is set to '19'.		
Configuration			
SIP Parameter	480: Reason: Q.850;cause=19 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name]		
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 19		
	[any boundary name]		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) → INVITE → 180 Ringing Start timer T9		
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	 Establish a communication from network A to network B, user B does not answer the communication setup. The ISUP timer T9 in the PSTN/PLMN expires Check: Is a CANCEL or BYE request is sent by the originating network? Check: Is an ISUP REL encapsulated in a BYE request? Check: Is the Cause value of the encapsulated REL set to '19'? Check: If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL? Check: Is a 487 Request Terminating send from the terminating user? Check: Are the media streams terminated after the 200 OK CANCEL/BYE was sent? NOTE: An ISUP REL is not encapsulated in a CANCEL request. Repeat this test in reverse direction. 		

7.1.5 Test purposes for Supplementary services

7.1.5.1 Test purposes for OIP

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

Test sees work on	00 sin 000		
Test case number	SS_oip_002		
Test case group	SIP-SIP/Service/OIP		
Reference	5.2.6.3/ [2]		
SELECTION EXPRESSION			
Test purpose	P-Preferred-Identity received, no match with the set of registered public identities. The terminating user receives the default public user identity of the originating user.		
	In case the preconditions are fulfilled to provide the terminating UE with originating identification information without preventing the presentation, ensure that an identity information in the P-Preferred-Identity header is provided by the originating UE, does not match with the set of registered public identities of the originating UE the terminating user receives a P-Asserted-Identity based on the default public user identity associated with the originating UE identifies the originator of the session.		
Configuration			
SIP Parameter	INVITE P-Asserted-Identity= default public user identity		
M	-Asserted-Identity- default public user identity		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → Apply post test routine		
Comments	 Check: Is the P-Asserted-Identity set to the default public user identity? Check: Is optional a second P-Asserted-Identity header present as a 'tel' URI with a public user identity? Check: If the user parameter is set to phone? Check: Is the P-Preferred-Identity header not present? Repeat this test in reverse direction. Repeat this test with all relevant end devices. 		

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

Test case number	SS_oip_(104		
	SIP-SIP/Service/OIP			
Test case group				
Reference		4.5.2.4/ [7]		
SELECTION EXPRESSION		ND NOT SE 19		
Test purpose	No Spec	ial arrangement exists.		
	information set of reg the AS se public us	sistered public identities of the origin ets the From header to the SIP URI er identity.	rmation in the From header with the nating user If is no match is found, that includes the registered default	
Configuration	Special a	rrangement for the originating user	does not exist	
SIP Parameter	INVITE			
		From=default public user identity		
	P-Asserted-Header=[any registered public user identity]			
Message flow				
SIP (Network A)		Interconnection Interface	SIP (Network B)	
		INVITE →		
		Apply post test routine		
Comments	Check:	Is the From header URI set to the URI?	value of the P-Asserted-Identity	
	Check:	Is the P-Asserted-Identity set to ar	ny registered public user identity?	
		Is the user parameter set to phone		
		nis test in reverse direction.		
	Repeat this test with all relevant end devices.			
L			-	

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

67

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

Test case number	SS_oip_(707	
Test case group	SIP-SIP/Service/OIP		
Reference	7.1.3/ [24]		
SELECTION EXPRESSION	[Network A] SE 17 [Network A] AND SE 47 AND SE 52 AND 52A		
Test purpose	SIP-I support. ISUP Additional Calling party number <i>presentation allowed</i> in the encapsulated IAM.		
	the BICC derived fr 'Presenta	when BICC/ISUP - SIP-I interworking applies in the originating network /ISUP IAM is encapsulated in the INVITE request. The From field is rom the Additional Calling party number in the encapsulated IAM. The ation restriction' indicator in the encapsulated IAM is set to 'allowed' no alue 'id' is present in the INVITE request.	
Configuration	The origin	nating user in the PSTN/PLMN part of Network A is subscribed to the ning option'	
SIP Parameter	INVITE From=[derived from the ISUP Additional calling party number] P-Asserted-Identity=[derived from the ISUP calling party number]		
		ontent-Type: multipart/mixed;boundary=[any boundary name]	
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	IAM		
	Calling party number		
	Screening indicator Network Provided		
		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		
Maaaaaa (law	[any boundary name]	
Message flow SIP (Network A)		Interconnection Interface SIP (Network B) INVITE(IAM) →	
Commonto	Check:	Apply post test routine Is a BICC/ISUP IAM encapsulated in the in the INVITE request?	
Comments	Check:	Is the Calling party number present in the encapsulated IAM and the screening indicator is set to 'Network Provided' and the Presentation	
	Check:	restriction indicator is set to 'allowed'? 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	
	Check:	indicator is set to 'allowed'? Is the From header field derived from the Additional calling party number in the encapsulated IAM?	
	Check: Repeat th	Is the value 'id' not present in the Privacy header field (if included)? his test in reverse direction.	

7.1.5.2 Test purposes for OIR

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

70

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

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

Test case number	SS_oir_0	04	
Test case group	SIP-SIP/Service/OIR		
Reference	7.1.3/ [24]		
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] SE 47 AND SE 52 AND 52A		
Test purpose	SIP-I support. ISUP Additional Calling party number <i>presentation restricted</i> in the encapsulated IAM.		
	the BICC/ derived fr 'Presenta	hen BICC/ISUP - SIP-I interworking applies in the originating network /ISUP IAM is encapsulated in the INVITE request. The From field is om the Additional Calling party number in the encapsulated IAM. The tion restriction' indicator in the Generic number parameter is set to no Privacy value 'id' is present in the INVITE request.	
Configuration	'no screer	nating user in the PSTN/PLMN part of Network A is subscribed to the ning option'	
SIP Parameter	Ino screening option' INVITE P-Asserted-Identity=[derived from the ISUP calling party number] From=[derived from the ISUP Additional calling party number] Privacy: id Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Calling party number Screening indicator Network Provided Presentation restriction restricted Address signal Generic number Number Qualifier Indicator Additional calling party number Screening indicator user provided, not verified Presentation restriction restricted		
	Address signal		
Message flow	[a	any boundary name]	
SIP (Network A)		Interconnection Interface SIP (Network B) INVITE(IAM) → Apply post test routine	
Comments	Check: Check:	Is a BICC/ISUP IAM encapsulated in the in the INVITE request? Is the Calling party number present in the encapsulated IAM and the screening indicator is set to 'Network Provided' and the Presentation	
	Check:	restriction indicator is set to 'restricted'? 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: Repeat th	Is the value 'id' present in the Privacy header field? is test in reverse direction.	

Test case number SS_tip_001 SIP-SIP/Service/TIP Test case group 5.2.6.4/ [8] Reference SELECTION EXPRESSION SE 21 Originating user receives the identity of the terminating user. Test purpose 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 B) SIP (Network A) Interconnection Interface → INVITE CASE A ← 180 Ringing CASE B ← 183 Session Progress CASE C ← 200 OK INVITE(P-Asserted-Identity) Apply post test routine Is the P-Asserted-Identity is present in a 180 Ringing or 183 Session Comments Check: Progress or in a 200 OK INVITE? Repeat this test in reverse direction.

Repeat this test with all relevant end devices.

73

7.1.5.3 Test purposes for TIP

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.			
rest purpose	Second identity provided in OF DATE.			
	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			
	Supported: from-change			
	P-Asserted-Identity:			
	UPDATE			
	From: (second user identity)			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE →			
	← 180 Ringing			
	← 200 OK INVITE(P-Asserted-Identity)			
	ACK →			
	← UPDATE (From)			
	200 OK UPDATE 🔶			
	Apply post test routine			
Comments	Check: Is the 'from-change' tag present in the Supported header of the initial			
	INVITE request?			
	Check: Is the P-Asserted-Identity 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 in reverse direction. Repeat this test with all chosen end devices.			

Test case number	SS tip 003		
	SIP-SIP/Service/TIP		
Test case group 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		
	not sent.		
Configuration	Special arrangement for the terminating user does not exist		
SIP Parameter	INVITE		
	Supported: from-change		
	200 OK INVITE		
	P-Asserted-Identity:		
	UPDATE		
	From: (default public user identity)		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE →		
	← 180 Ringing		
	 200 OK INVITE(P-Asserted-Identity) 		
	ACK →		
	Apply post test routine		
Comments	Check: Is the 'from-change' tag present in the Supported header of the initial		
	INVITE request?		
	Check: Is the P-Asserted-Identity present in the 200 OK INVITE?		
	Check: Is the 'from-change' tag present in the supported header of the		
	provisional (18x) or final (200 OK) response?		
	Check: Is no UPDATE request sent by the terminating		
	Repeat this test in reverse direction.		
	Repeat this test with all relevant end devices.		

Test case number	SS_tip_00	4			
Test case group		SIP-SIP/Service/TIP			
Reference	6.7/ [24]				
SELECTION EXPRESSION		3] SE 17 AND [Network A] SE 4	7 AND SE 53		
Test purpose					
		SIP-I support. The Connected number presentation allowed is present in the encapsulated 200 OK.			
	line enear				
	Ensure the	Ensure that on receipt of a 200 OK INVITE to establish a confirmed dialogue an			
		s encapsulated if SIP-I - BICC/ISUP interworking is applicable in Network			
		e Address presentation restriction indicator is set to 'allowed'. The			
	screening	indicator is set to Network provi	ded or user provided, verified and		
	passed.				
Configuration					
SIP Parameter	200 OK IN				
	Co	ntent-Type: multipart/mixed;bou	ndary=[any boundary name]		
		ny boundary name]			
		ntent-Type: application/isup;vers			
	Co	ntent-Disposition: signal;handlin	g=required		
		A NIR.			
		ANM			
		Connected number			
		Screening indicator Network provided or user provided, verified and passed			
		Address presentation restriction			
		allowed			
		Address signal			
	[a	[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			lated 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 th	s test in reverse direction.			

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

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

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

Test case number	SS_tir_0	01B		
Test case group		Service/TIR		
Reference	4.5.2.7/ [9]			
SELECTION EXPRESSION	SE 23	<u>.</u>		
Test purpose	CDIV occurs. Originating user does not receive the identity of the diverted			
	to user.	5 5	,	
	Ensure th	nat, when Call diversion occurs, the	e identity of the diverted-to user is	
	restricted	I when the diverted-to user is subs	cribed to the TIR service and requires	
		nt the presentation of his/her identit		
			e 180 or 200 OK INVITE identifying	
		ted-to user contains an escaped 'P		
Configuration		rted-to user subscribes to the 'TIR'	service	
SIP Parameter	180/200			
	History-Ir			
	<sip:userb@networkb>;index=1,</sip:userb@networkb>			
	<sip: userc@networkb;cause="[any]?Privacy=history">;index=1.1</sip:>			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	=(.)			
	← INVITE(2) 180 Ringing(2) →			
	←			
	•	200 OK INVITE(2)	→	
	← ACK			
	← 200 OK INVITE(1)			
	Apply post test routine			
Comments	Check:			
	originating user?			
	Check:	neck: Is the Privacy header is escaped in the hi-entry identify the diverted-to		
		user set to 'history'?		
	Repeat this test in reverse direction.			
	Repeat the	nis test with all chosen end devices	5.	

Test case number	SS_tir_002			
Test case group	SIP-SIP/Service/TIR			
Reference	6.7/ [24]			
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 53			
Test purpose	SIP-I support. The Connected number presentation allowed is present in the encapsulated 200 OK.			
	uie envapoulateu 200 UN.			
	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 networking			
	B. The Address presentation restriction indicator is set to 'restricted'. The			
	screening indicator is set to 'Network provided' or 'user provided, verified and			
	passed'.			
Configuration				
SIP Parameter	200 OK INVITE			
	Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name]			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	A > 104			
		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		SIP-SIP/Service/TIR			
Reference	6.7/ [24]				
SELECTION EXPRESSION		[Network B] SE 17 AND [Network B] SE 47 AND SE 53 AND SE 53A			
Test purpose	SIP-I sup	SIP-I support. The additional connected number restricted is present in the encapsulated 200 OK.			
	ANM is e B. A Gen 'additiona not verifie A Connec	Ensure that on receipt of a 200 OK INVITE to establish a confirmed dialogue an ANM is encapsulated if SIP-I - BICC/ISUP interworking is applicable in Network B. A Generic number parameter is present the Number qualifier indicator set to 'additional connected number' the Screening indicator is set to 'user provided, not verified' and the Address Presentation Restricted is set to 'restricted'. A Connected number parameter is present the Screening indicator is set to 'Ivetwork provided' and the Address Presentation Restricted indicator is set to 'Ivetwork provided' and the Address Presentation Restricted indicator is set to 'Ivetwork provided' and the Address Presentation Restricted indicator is set to 'Ivetwork' provided' and the Address Presentation Restricted indicator is set to 'Ivetwork' provided' and the Address Presentation Restricted indicator is set to 'Ivetwork' provided' and the Address Presentation Restricted indicator is set to 'Ivetwork' provided' and the Address Presentation Restricted indicator is set to 'Ivetwork' provided' and the Address Presentation Restricted indicator is set to 'Ivetwork' provided' and the Address Presentation Restricted indicator is set to 'Ivetwork' provided' and the Address Presentation Restricted indicator is set to 'Ivetwork' provided' and the Address Presentation' Prese			
Configuration	The term		rt of network B is subscribed to the		
SIP Parameter	200 OK II P-	200 OK INVITE P-Asserted-Identity=[derived from the ISUP Connected number] Content-Type: multipart/mixed;boundary=[any boundary name]			
	Co	any boundary name] ontent-Type: application/isup;versic ontent-Disposition: signal;handling=			
		ANM			
		Connected number			
		Screening indicator			
		Network provided or user provided, verified and			
		passed			
		Presentation restriction			
		restricted			
		Address signal			
		Generic number			
		Number Qualifier Indicator			
		Additional connected	number		
		Screening indicator			
	user provided, not verified				
	Address Presentation Restricted				
	restricted				
	Address signal				
	[[any boundary name]			
Message flow					
SIP (Network A)		Interconnection Interface	SIP (Network B)		
		INVITE(IAM)	→		
	+ +	180 Ringing(ACM)			
	~	200 OK INVITE(ANM) ACK	→		
		Apply post test routine	7		
Comments	Check:	Is the BICC/ISUP ANM encapsula	ated in the 200 OK INVITE final		
	•	response?			
	Check:				
	Check:	Is the Number Qualifier Indicator			
		'additional connected number'?			
	Check:		Generic number set to 'user provided,		
	not verified'?				
	Check: Is the Address presentation restriction indicator in the Generic number 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/HOLD		
Reference	4.5.2.1/[17]		
SELECTION EXPRESSION	SE 24		
Test purpose	Hold the session the media stream was previously set to sendrecv.		
	Ensure that the UE A requesting hold of the session sends an INVITE or UPDATE request to hold the session. Hold is done containing the SDP with the attribute "a=sendonly". The UE A after requesting the hold session <i>receives</i> 200 OK final response containing the SDP with the attribute "a=recvonly".		
Configuration			
SIP Parameter			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) A confirmed session already exists		
CASE A	INVITE(sendonly) → ← 200 OK INVITE (recvonly) ACK →		
CASE B	UPDATE(sendonly) → ← 200 OK UPDATE (recvonly) Apply post test routine		
Comments	Check: Is the user in network A able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?		
	Repeat this test in reverse direction.		

82

Test case number	SS_hold_002			
Test case group		Service/HOLD		
Reference	4.5.2.1/ [17]			
SELECTION EXPRESSION	SE 24			
Test purpose	Hold the session the media stream was previously set to recvonly.			
		nat the UE B requesting <mark>hold</mark> of the		
		INVITE or UPDATE request to hold		
		g the SDP with the attribute " <mark>a=sen</mark>		
		A after requesting to hold the held se		
O fi	UPDATE	request containing the SDP with th	ie attribute "a= <mark>inactive</mark> ".	
Configuration				
SIP Parameter				
Message flow		Interconnection Interface	SID (Network B)	
SIP (Network A)	۸	Interconnection Interface	SIP (Network B)	
CASE A	↔ A C	onfirmed session already exists INVITE (sendonly)		
	•	200 OK INVITE (recvonly)	→	
	←	ACK	2	
	•	INVITE (<mark>inactive</mark>)	→	
	+	200 OK INVITE (inactive)		
		ACK	→	
CASE B	+	INVITE (<mark>sendonly</mark>)	_	
		200 OK INVITE (recvonly)	→	
	+			
	←	UPDATE(<mark>inactive</mark>) 200 OK UPDATE (inactive)	→	
	T	200 OK UPDATE (Inactive)		
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)	→	
			→	
	+	200 OK UPDATE (inactive)		
Commonto	Cheek	Apply post test routine	t the appaien on hold by conding on	
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 set the session on hold by sending an			
	Oneon.	INVITE or UPDATE request and the version parameter in the SDP 'o'		
		line is incremented?		
	Repeat t	his test in reverse direction.		

Test case number	SS_hold	003	
Test case group			
Reference	4.5.2.1/[17]	
SELECTION EXPRESSION	SE 24	3	
Test purpose	Resume the session the media stream was previously set to sendonly.		as previously set to sendonly.
	Ensure that the UE A is requested to resume the session with user B the UE-A starts sending media and sends an INVITE or UPDATE request to resume the session with the attribute "a=sendrecv in the SDP. The UE A after requesting to resume the held session <i>receives</i> 200 OK final response and optionally the attribute "a=sendrecv in the SDP. The a=sendrecv attribute is the default value therefore the attribute can be omitted.		
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
	A co	onfirmed session already exists	_
CASE A	-	INVITE (<mark>sendonly</mark>)	→
	←	200 OK INVITE (recvonly)	
			→
	←	INVITE (<mark>sendrecv</mark>)	7
	~	200 OK INVITE (sendrecv) ACK	→
		ACK	7
CASE B		UPDATE (<mark>sendonly</mark>)	→
CASE B	←	200 OK UPDATE (recvonly)	
	•	UPDATE (sendrecv)	→
	←	200 OK UPDATE (sendrecv)	2
	-	Apply post test routine	
Comments	Check:		et the session on hold by sending an
			the version parameter in the SDP 'o'
		line is incremented?	
	Check:	Is the user in network A able to re	etrieve the session by sending an
			the version parameter in the SDP 'o'
		line is incremented? The absence	e of the 'sendrecv' attribute is the
		default value.	
	Repeat t	his test in reverse direction.	

Test case number	SS_hold	004	
Test case group		Service/HOLD	
Reference	4.5.2.1/ [
SELECTION EXPRESSION	SE 24		
		the cossion the modio stream we	a proviously set to inactive
Test purpose	Resume	the session the media stream wa	is previously set to mactive.
	The See	nion in the "inactive" state. Ensure	that the LIF A is requesting to
		sion is in the "inactive" state. Ensure	
		he session with user B the UE-A ser	
		he session with the attribute <mark>"a=rec\</mark>	
		ng to resume the held session receiv	es 200 OK final response with the
	attribute	"a=sendonly in the SDP.	
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
	Ac	onfirmed session already exists	
CASE A	←	INVITE(sendonly)	
		200 OK INVITE (recvonly)	→
	←	ACK	
		INVITE(<mark>inactive</mark>)	→
	←	200 OK INVITE (inactive)	
	-	ACK	→
		INVITE (recvonly)	→
	←	200 OK INVITE (sendonly)	-
	*	ACK	→
		AON	
CASE B	←	INI) /ITE (condonly)	
CASE D	T	INVITE(sendonly)	→
		200 OK INVITE (recvonly)	7
	÷	ACK	
	-	UPDATE(inactive)	→
	+	200 OK UPDATE (inactive)	_
	_	INVITE (<mark>recvonly</mark>)	→
	←	200 OK (<mark>sendonly</mark>)	
		ACK	→
CASE C	÷	UPDATE (sendonly)	
		200 OK UPDATE (recvonly)	→
		INVITE(<mark>inactive</mark>)	→
	÷	200 OK INVITE (inactive)	
		ACK	→
		UPDATE (<mark>recvonly</mark>)	→
	←	200 OK UPDATE (<mark>sendonly</mark>)	
CASE D	←	UPDATE (sendonly)	
		200 OK UPDATE (recvonly)	→
		UPDATE(inactive)	→
	←	200 OK UPDATE (inactive)	-
	-	UPDATE (recvonly)	+
	←	200 OK UPDATE (sendonly)	2
	•	Apply post test routine	
Comments	Check:		t the session on hold by sending an
Comments	Check:		ne version parameter in the SDP 'o'
			e version parameter in the SDP 0
	Charles	line is incremented?	t the exercise on hold by conding an
	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?	
	Repeat t	his test in reverse direction.	

Test case number	SS_hold	I_005	
Test case group	SIP-SIP/	/Service/HOLD	
Reference	4.5.2.1/	[17]	
SELECTION EXPRESSION	SE 24		
Test purpose	Hold the session the media stream was previously set to sendrecv.		
	Ensure t	hat the UE B sends an INVITE or U	PDATE request to hold the session.
		done containing the SDP with the att	
	sends a	200 OK final response containing th	ne SDP with the attribute
	"a= <mark>recvo</mark>	only" and stops sending media.	
Configuration			
SIP Parameter			
Message flow SIP (Network A)		Interconnection Interface	SIP (Network B)
	Ac	onfirmed session already exists	··· (
CASE A	+	INVITE(sendonly)	
		200 OK INVITE(recvonly)	→
	←	ACK	
CASE B	←	UPDATE(<mark>sendonly</mark>)	
		200 OK UPDATE (recvonly)	→
		Apply post test routine	
Comments	Check:		et the session on hold by sending an
			he version parameter in the SDP 'o'
		line is incremented?	
	Repeat t	this test in reverse direction.	

Test case number	SS_hold	_006	
Test case group	SIP-SIP/Service/HOLD		
Reference	4.5.2.1/ [17]		
SELECTION EXPRESSION	SE 24		
Test purpose	Hold the	e session the media stream was p	reviously set to sendonly.
		·	
		sion is in the held state done by UE-	
		or UPDATE request to hold the sess	
		n the attribute "a= <mark>inactive</mark> ". The UE /	
		00 OK final response containing the	SDP with the attribute "a=inactive"
0	and stop	s sending media.	
Configuration	_		
SIP Parameter			
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
CASE A	AC	onfirmed session already exists INVITE(<mark>sendonly</mark>)	→
CASEA	←	200 OK INVITE (recvonly)	7
	L	ACK	→
	←	INVITE (inactive)	2
	-	200 OK INVITE (inactive)	→
	+	ACK	
CASE B		INVITE(<mark>sendonly</mark>)	→
	÷	200 OK INVITE (recvonly)	
	_	ACK	→
	+	UPDATE (inactive)	
		200 OK UPDATE (<mark>inactive</mark>)	→
CASE C		UPDATE (<mark>sendonly</mark>)	→
CASE C	←	200 OK UPDATE (recvonly)	7
	÷	INVITE (inactive)	
	-	200 OK INVITE (inactive)	→
	+	ACK	
CASE D		UPDATE (<mark>sendonly</mark>)	→
	÷	200 OK UPDATE (recvonly)	
	÷	UPDATE (inactive)	
		200 OK UPDATE (<mark>inactive</mark>)	→
- · · ·		Apply post test routine	
Comments	Check:		t the session on hold by sending an
			he version parameter in the SDP 'o'
	Check:	line is incremented?	t the session on hold by sending an
	CHECK:		he version parameter in the SDP 'o'
		line is incremented?	
	Repeat t	his test in reverse direction.	
	ricpedit		

Test case number	SS_hold	_007	
Test case group	SIP-SIP/	Service/HOLD	
Reference	4.5.2.1/[17]	
SELECTION EXPRESSION	SE 24	-	
Test purpose	Resume the session the media stream was previously set to recvonly.		as previously set to recvonly.
		nat the UE B sends an INVITE or U	
			tarts sending media. Resume is done
			ndrecv". The UE A after receiving the
	Resume	of the session <i>sends</i> 200 OK final rute "a=sendrecv". The a=sendrecv	response containing the SDP with
		the attribute can be omitted.	attribute is the default value
Configuration	literetore	the attribute can be offitted.	
SIP Parameter			
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
	A co	onfirmed session already exists	••• (••••••=)
CASE A	÷	INVITE (sendonly)	
		200 OK INVITE(<mark>recvonly</mark>)	→
	÷	ACK	
	÷	INVITE(sendrecv)	
	-	200 OK INVITE(<mark>sendrecv</mark>)	→
	÷	ACK	
CASE B	←	UPDATE (sendonly)	
		200 OK UPDATE (recvonly)	→
	÷	UPDATE (sendrecv)	
		200 OK UPDATE (<mark>sendrecv</mark>)	→
	1	Apply post test routine	
Comments	Check:		et the session on hold by sending an
			he version parameter in the SDP 'o'
	Check:	line is incremented?	triove the exercise by conding on
	Check:	Is the user in network B able to re	he version parameter in the SDP 'o'
		line is incremented?	
	Repeat th	his test in reverse direction.	
	1. top cold		

Test case number	SS_hold	008	
Test case group		_000 Service/HOLD	
Reference	4.5.2.1/[
SELECTION EXPRESSION	SE 24		
Test purpose		the session the media stream wa	s proviously set to inactive
rest purpose	Resume	the session the media stream wa	is previously set to mactive.
	The See	sion is in the "inactive" state. Ensure	that the LIE R cando on INIVITE or
			session with user A, the UE-A starts
		media. Resume is done containing t	esume of the session sends 200 OK
Configuration	linai resp	oonse containing the SDP with the a	illindule a= <mark>sendoniy</mark> .
Configuration			
SIP Parameter			
Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
	AC	onfirmed session already exists	
CASE A		INVITE (sendonly)	→
	÷	200 OK INVITE (recvonly)	
		ACK	→
	←	INVITE (<mark>inactive</mark>)	
		200 OK INVITE (inactive)	→
	÷	ACK	
	÷	INVITE (<mark>recvonly</mark>)	
		200 OK INVITE (<mark>sendonly</mark>)	→
	÷	ACK	
CASE B	,	INVITE (sendonly)	→
	÷	200 OK INVITE (recvonly)	
			→
	÷	UPDATE (inactive)	-
	L	200 OK UPDATE (inactive)	→
	÷		→
		200 OK UPDATE (<mark>sendonly</mark>)	7
CASE C		UPDATE (sendonly)	→
	←	200 OK UPDATE (recvonly)	
	÷	INVITE (inactive)	
	N	200 OK INVITE (inactive)	→
	←	ACK	2
	÷	INVITE (recvonly)	
	•	200 OK INVITE (sendonly)	→
	←	ACK	2
	•	//or	
CASE D		UPDATE (sendonly)	→
	←	200 OK UPDATE (recvonly)	
	÷	UPDATE (inactive)	
	-	200 OK UPDATE (inactive)	→
	←	UPDATE (recvonly)	_
		200 OK UPDATE (sendonly)	→
		Apply post test routine	
Comments	Check:		t the session on hold by sending an
			ne version parameter in the SDP 'o'
	1	line is incremented?	· · · · · · · · · · · · · · · · · · ·
	Check:		t the session on hold by sending an
			ne version parameter in the SDP 'o'
		line is incremented?	
	Check:	Is the user in network B able to ref	trieve the session by sending an
			ne version parameter in the SDP 'o'
	1	line is incremented?	
	Repeat t	his test in reverse direction.	
	1.1.50.500.1		

Test case number	SS_hold_009	
Test case group	SIP-SIP/Service/HOLD	
Reference	4.5.2.1/[17]	
SELECTION EXPRESSION	SE 24	
Test purpose	Resume the session on both sides the me	dia stream was previously set to
	inactive.	
	The Session is in the "inactive" state. Ensure	that the UE A is requesting to
	resume the session with user B, the UE-A sta	
	INVITE or UPDATE request to resume the se	
	"a= recvonly in the SDP.	
	The UE A after requests to resume the sessi	on receives 200 OK final response
	containing the SDP with the attribute "a=sen	
	The UE B after requests to resume the sessi	
	request containing the SDP with the attribute	
	attribute is the default value therefore the attribute	ribute can be omitted.
Message flow		
SIP (Network A)	Interconnection Interface	SIP (Network B)
	A confirmed session already exists	х <i>у</i>
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	→
	INVITE(sendrecv)	
	200 OK INVITE (sendrecv)	→
	← ACK	
		_
CASE B	INVITE(sendonly)	→
	← 200 OK INVITE (recvonly)	
	ACK	→
	← UPDATE (inactive)	
	200 OK UPDATE (inactive)	→
	INVITE(recvonly)	→
	← 200 OK INVITE (sendonly)	
		→
	← UPDATE (sendrecv)	→
	200 OK UPDATE (sendrecv)	7
CASE C	UPDATE (sendonly)	→
CASEC	← 200 OK UPDATE (recvonly)	7
	← INVITE(inactive)	
	200 OK INVITE (inactive)	→
	← ACK	
	UPDATE (recvonly)	→
	← 200 OK UPDATE (sendonly)	2
	ACK	→
	← INVITE(sendrecv)	2
	200 OK INVITE (sendrecv)	→
	← ACK	2
CASE D	UPDATE (sendonly)	→
	← 200 OK UPDATE (recvonly)	
	← UPDATE (inactive)	
	200 OK UPDATE (inactive)	→
	UPDATE (recvonly)	→
	← 200 OK UPDATE (sendonly)	
	← UPDATE (sendrecv)	
	200 OK UPDATE (sendrecv)	→
	Apply post test routine	
L	FILLY PETERSON	

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?
		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 t	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 mo	edia stream was previously set to
rest purpose	The Session is in the "inactive" state. Ensure UPDATE request to resume the session with media. Resume is done containing the SDP The UE A after receiving the Resume of the response containing the SDP with the attribut The UE A after requests to resume the sess request containing the SDP with the attribute The UE B after receiving the Resume of the	e that the UE B sends an INVITE or n user A, the UE-B starts sending with the attribute "a= <mark>recvonly</mark> ". session <i>sends</i> 200 OK final ute "a= <mark>sendonly</mark> ". ion <i>sends an INVITE or UPDATE</i> e "a= <mark>sendrecv</mark> .
	response containing the SDP with the attribute is the default value therefore the att	ute "a= <mark>sendrecv</mark> ". The a=sendrecv
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 INVITE(inactive)	→
	← 200 OK INVITE (inactive) ACK	→
	 INVITE(recvonly) 200 OK INVITE (sendonly) 	→
	← ACK INVITE(<mark>sendrecv</mark>)	→
	← 200 OK INVITE (sendrecv) ACK	→
CASE B	← INVITE(sendonly)	`
	← 200 OK INVITE (recvonly) ← ACK	→ 、
	UPDATE (inactive) ← 200 OK UPDATE (inactive) ← INVITE(recvonly)	→
	200 OK INVITE (sendonly) ← ACK	→
	UPDATE (sendrecv) COU OK UPDATE (sendrecv)	→
CASE C	← UPDATE (sendonly) 200 OK UPDATE (recvonly) INVITE(inactive)	→ →
	← 200 OK INVÌTE (<mark>inactive</mark>) ACK	→
	← UPDATE (recvonly) 200 OK UPDATE (sendonly) INVITE(sendrecv)	→ →
	← 200 OK INVITE (sendrecv) ACK	→
CASE D	 UPDATE (sendonly) 200 OK UPDATE (recvonly) UPDATE (inactive) 	→ →
	 ← 200 OK UPDATE (inactive) ← UPDATE (recvonly) 200 OK UPDATE (sendonly) 	→
	UPDATE (sendrecv) COUCK (sendrecv) Apply post test routine	→

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

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

Test case number	SS_hold_013			
Test case group	SIP-SIP/Service/HOLD			
Reference	B.10/ [24]			
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] SE 47 AND SE 54			
Test purpose	SIP-I support. Hold requested by the originating user, Hold by the terminating user. Retrieve requested by the originating user.			
	 Ensure the hold and retrieve procedure when ISUP - SIP-I interworking applies in the network A: Originating user in network A places the session on hold. 			
	 Terminating user in network B places the session on hold. 			
	Originating user in network A retrieves the session.			
	 Terminating user in network B retrieves the session. 			
	Verify the Generic notification parameter in the encapsulated CPG present in the			
SIP Parameter	INVITE request from the network A. 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			
	← ACK			
	INVITE(<mark>sendonly</mark> , CPG <mark>retrieval</mark>) →			
	← 200 OK INVITE (recvonly)			
	ACK →			
	← INVITE(sendrecv)			
	200 OK INVITE (sendrecv) →			
	← ACK			
	Apply post test routine			
Comments	Establish a session from network A to network B			
	The user in the PSTN/PLMN part of Network A places the session on hold. Check: Is a CPG encapsulated in the INVITE request?			
	Check: Is a Generic notification parameter present the Notification indicator			
	set to 'remote hold'?			
	Check: Is the 'a' attribute in the SDP set to 'sendonly'?			
	Check: Is the Version parameter in the SDP incremented?			
	The user in Network B places the session on hold Check: Is the 'a' attribute in the SDP set to 'inactive'?			
	Check: Is the Version parameter in the SDP incremented?			
	The user in Network A retrieves the session			
	Check: Is a CPG encapsulated in the INVITE request?			
	Check: Is a Generic notification parameter present the Notification indicator			
	set to ' remote retrieval '? Check: Is the 'a' attribute in the SDP set to ' sendonly '?			
	Check: Is the Version parameter in the SDP set to sendomy ?			
	The user in Network B retrieves the session			
	Check: Is the 'a' attribute in the SDP set to 'sendrecv'?			
	Check: Is the Version parameter in the SDP incremented?			
	Repeat this test in reverse direction.			

Test case number	SS_hold_014			
Test case group	SIP-SIP/Service/HOLD			
Reference	B.10/ [24]			
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] SE 47 AND SE 54			
Test purpose	 SIP-I support. Hold requested by the originating user, Hold by the terminating user. Retrieve requested by the terminating user. Ensure the hold and retrieve procedure when ISUP - SIP-I interworking applies in the network A: Originating user in network A places the session on hold. Terminating user in network B places the session on hold. Terminating user in network B retrieves the session. Originating user in network A retrieves the session. 			
	Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the network A.			
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) INVITE (sendonly, CPG hold) → 200 OK INVITE (recvonly) ACK			
	200 OK INVITE (inactive) → ← ACK			
	← INVITE(recvonly)			
	200 OK INVITE (sendonly) → ← ACK			
	INVITE(sendrecv, CPG retrieval) → ← 200 OK INVITE (sendrecv)			
	ACK →			
	Apply post test routine			
Comments	Establish a session from network A to network B The user in the PSTN/PLMN part of Network A places the session on hold.			
	Check: Is a CPG encapsulated in the INVITE request?			
	Check: Is a Generic notification parameter present the Notification indicator			
	set to 'remote hold'?			
	Check: Is the 'a' attribute in the SDP set to 'sendonly'?			
	Check: Is the Version parameter in the SDP incremented?			
	The user in network B places the session on hold			
	Check: Is the 'a' attribute in the SDP set to 'inactive'? Check: Is the Version parameter in the SDP incremented?			
	The user in network B retrieves the session			
	Check: Is the 'a' attribute in the SDP set to 'recvonly'?			
	Check: Is the Version parameter in the SDP incremented?			
	The user in network A retrieves the session			
	Check: Is a CPG encapsulated in the INVITE request? Check: Is a Generic notification parameter present the Notification indicator			
	set to 'remote retrieval'?			
	Check: Is the 'a' attribute in the SDP set to 'sendrecv'?Check: Is the Version parameter in the SDP incremented?			
	Repeat this test in reverse direction.			

Test case number	SS_hold_015			
Test case group	SIP-SIP/Service/HOLD			
Reference	B.10/ [24]			
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] SE 47 AND SE 54			
Test purpose	SIP-I support. Hold requested by the terminating user, Hold by the originating user. Retrieve requested by the originating user. Ensure the hold and retrieve procedure when ISUP - SIP-I interworking applies			
	 in the Network A: Terminating user in Network B places the session on hold. Originating user in Network A places the session on hold. Originating user in Network A retrieves the session. Terminating user in Network B retrieves the session. 			
	Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A.			
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			
	Generic notification			
	remote hold			
	or			
	remote retrieval			
	[any boundary name]			
Message flow	Interconnection Interface CID (Network D)			
SIP (Network A)	Interconnection Interface SIP (Network B) A confirmed session already exists			
	← INVITE(sendonly)			
	200 OK INVITE (recvonly) →			
	← ACK			
	INVITE(<mark>inactive</mark> , CPG <u>hold</u>) → ← 200 OK INVITE (inactive)			
	← 200 OK INVITE (inactive) ACK →			
	INVITE(recvonly, CPG retrieval) →			
	← 200 OK INVITE (sendonly)			
	ACK →			
	← INVITE(sendrecv)			
	200 OK INVITE (sendrecv) → ← ACK			
	Apply post test routine			
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?			
	Check: Is the Version parameter in the SDP incremented? The user in Network A places the session on hold			
	Check: Is a CPG encapsulated in the INVITE request?			
	Check: Is a Generic notification parameter present the Notification indicator			
	set to 'remote hold'?			
	Check: Is the 'a' attribute in the SDP set to 'inactive'?			
	Check: Is the Version parameter in the SDP incremented?			
	The user in Network A retrieves the session Check: Is a CPG encapsulated in the INVITE request?			
	Check: Is a Generic notification parameter present the Notification indicator			
	set to 'remote retrieval'?			
	Check: Is the 'a' attribute in the SDP set to 'recvonly'?			
	Check: Is the Version parameter in the SDP incremented?			
	The user in Network B retrieves the session			
	Check: Is the 'a' attribute in the SDP set to 'sendrecv'?			
	Check: Is the Version parameter in the SDP incremented? Repeat this test in reverse direction.			
	וופטנוטוו.			

Test case number	SS_hold_016			
Test case group	SIP-SIP/Service/HOLD			
Reference	B.10/ [24]			
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] 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 INVITE request from the Network A. 			
SIP Parameter	INVITE:			
	Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name]			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required CPG			
	Generic notification			
	remote hold			
	or			
	remote retrieval			
	[any boundary name]			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) A confirmed session already exists			
	← INVITE(sendonly)			
	200 OK INVITE (recvonly) →			
	← ACK			
	INVITE(<mark>inactive</mark> , CPG hold) →			
	 € 200 OK INVITE (inactive) ACK → 			
	← INVITE(recvonly) 200 OK INVITE (sendonly) →			
	← ACK			
	INVITE(sendrecv, CPG retrieval) → Control Control Co			
	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 'sendrecv'? 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		Service/CFU			
Reference	4.5.2.6/ [
SELECTION EXPRESSION	SE 25	5			
Test purpose		nication forwarding uncondition	al basic r	ules	
rest purpose	Commu		iai, basic i	ules.	
	The user	A and user C are in network A. Th	na usar R i	s in network B and is	
		with CFU.			
		hat when user A calls user B, the c	call is forwa	arded unconditional to user	
		active call state, ensure the prope			
Configuration					
SIP Parameter					
Message flow					
SIP (Network A)		Interconnection Interface		SIP (Network B)	
		INVITE(Call-ID A-B)	→		
		CFU is performed			
	←	INVITE(Call-ID B-C)			
		180 Ringing(Call-ID C-B)	→		
	←	180 Ringing (Call-ID B-A)			
		200 OK INVITE(Call-ID C-B)	→		
	←	ACK(Call-ID B-C)			
	←	200 OK INVITE(Call-ID B-A)			
		ACK(Call-ID A-B)	→		
		Communication			
		Apply post test routine			
Comments	Check:	CDIV unconditional is successful			
	Check:	In the active call state, ensure the			
	Check: Is the P-Asserted-Identity present in the INVITE sent from networ				
		to network A set to the identity of	f the origin	ating user?	
	Repeat t	Repeat this test in reverse direction.			

Toot oooo number	CC of u	000			
Test case number	SS_cfu_				
Test case group	·····	Service/CFU			
Reference	4.5.2.6/ [9]			
SELECTION EXPRESSION	SE 25 AI	ND SE 30			
Test purpose	Commu	nication forwarding uncondition	onal, no noti	fication.	
	provided his comn Ensure ti	The user A and user C are in network A. The user B is in network B and is provided with CFU, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B, the call is forwarded unconditional to user C, the originating user is not notified.			
Configuration	Subscri	Subscription options: Originating user receives notification that his communication has been diverted =			
	No No	ig user receives nouncation that		ication has been diverted =	
SIP Parameter					
Message flow SIP (Network A)		Interconnection Interface		SIP (Network B)	
		INVITE(Call-ID A-B)	→		
	+	CFU is performed INVITE(Call-ID B-C)			
		180 Ringing(Call-ID C-B)	→		
	+	180 Ringing(Call-ID B-A)			
		Apply post test routine			
Comments	Check:	No notification regarding call fo interconnection interface.	rwarding in r	network B is received at the	
	Repeat t	his test in reverse direction.			

Test case number	SS_cfu_00	03			
Test case group		ervice/CFU			
Reference	4.5.2.6/ [9]				
SELECTION EXPRESSION		SE 25 AND SE 30			
Test purpose		Communication forwarding unconditional, originating user is notified. UR			
l'est puipose		erted-to user not received.			
	or the dive				
	provided w has been o to URI to o the presen Ensure tha C, user A i and served	A and user C are in network A. The user B is in network B and is with CFU "Originating user receives notification that his communication diverted" = Yes and "Served user allows the presentation of forwarded originating user in diversion notification" = No and. "Served user allows nation of his/her URI to originating user in diversion notification" = No. at when user A calls user B, the call is forwarded unconditional to user is notified of call diversion and not informed of the diverted-to number d user number.			
Configuration		tion options:			
		Driginating user receives notification that his communication has been			
		liverted = <mark>Yes</mark>			
		Served user allows the presentation of forwarded to URI to originating			
		user in diversion notification = <mark>No</mark>			
		Served user allows the presentation of his/her URI to originating user in liversion notification = No			
SIP Parameter					
SIP Parameter		181 Being Forwarded			
		P-Asserted-Identity: <userb@networkb> Privacy: id History-Info:</userb@networkb>			
		<pre><sip:userb@networkb?privacy=history>;index=1,</sip:userb@networkb?privacy=history></pre>			
		p: userC@networkA;cause=302 <mark>?Privacy=history</mark> >;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)			
	← 18	31 Being Forwarded (Call-ID B-A)			
		Apply post test routine			
Comments		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'.			
		Is the cause parameter in the last entry is set to '302'?			
		Is the "user=phone" parameter present in all History-Info header URIs?			
	Check:	Is the P-Asserted-Identity header present in the 181 identifying the served user?			
		The history entries can be accumulated in "one" History-Info header			
		or each history entry is present in one single History-Info header.			
	Repeat thi	is test in reverse direction.			

Test case number	SS_cfu_0	004				
Test case group		Service/CFU				
Reference	4.5.2.6/ [9]					
SELECTION EXPRESSION	SE 25 AND SE 30					
Test purpose	Communication forwarding unconditional, originating user is notified. from the diverted-to user received.					
	provided has been to URI to Ensure th	A and user C are in network 1. The user B is in network N2 and is with CFU "Originating user receives notification that his communication diverted" = Yes and "Served user allows the presentation of forwarded originating user in diversion notification" = Yes. Net when user A calls user B, the call is forwarded unconditional to user is notified of call diversion and informed of the diverted-to number.				
Configuration		tion 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 = <mark>Yes</mark>				
SIP Parameter		g Forwarded				
	P-Asserted-Identity: <userb@networkb></userb@networkb>					
	History-Info:					
		<sip:userb@networkb>;index=1,</sip:userb@networkb>				
		ip: 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				
	← ← 1	INVITE(Call-ID B-C) 81 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'?				
	Check:	Is the "user=phone" parameter present in all History-Info header URIs?				
	Check:	Is the P-Asserted-Identity header present in the 181 identifying the served user?				
	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 th	his 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				
lest pulpose	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 u	ser			
	C, user C is not informed of the forwarding number.				
Configuration	Subscription options:				
5	 Served user allows the presentation of his/her URI to the diverted-to 	2			
	user = <mark>No</mark>	-			
SIP Parameter					
	History-Info:				
	<pre><sip:userb@networkb?privacy=history< pre="">;index=1,</sip:userb@networkb?privacy=history<></pre>				
	<sip: userc@networka;cause="302">;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)				
	Apply post test routine				
Comments	Check: A History-Info header is received in the INVITE contains the URI o				
	user B (served user) at the interconnection interface and a Privacy	/			
	header is escaped set to 'history'.				
	Check: Is the cause parameter in the last entry is set to '302'?				
	Check: Is the "user=phone" parameter present in all History-Info header URIs?				
	NOTE: The history entries can be accumulated in "one" History-Info head	or			
	or each history entry is present in one single History-Info header.	C1			
	NOTE: The Request line may contain a 'cause' parameter indicating the				
	redirecting reason.				
	Repeat this test in reverse direction.				

Test case number	SS cfu (006				
Test case group		Service/CFU				
Reference	4.5.2.6/ [
SELECTION EXPRESSION	SE 25 AN					
		nication forwarding unconditional, diverted-to user receives the				
Test purpose		e served user.				
		le serveu user.				
	The user	A and user C are in network A. The user B is in network B and is				
		with CFU "Served user allows the presentation of his/her URI to				
		to user" = Yes.				
		hat when user A calls user B, the call is forwarded unconditional to user				
		is informed of the forwarding number.				
Configuration		otion options:				
3 1 1 1		Served user allows the presentation of his/her URI to diverted-to user =				
		Yes				
SIP Parameter	INVITE:					
	History-Ir	nfo				
		<sip:userb@networkb>;index=1,</sip:userb@networkb>				
	<sip:< th=""><th colspan="4"><sip: userc@networka;cause="302">;index=1.1</sip:></th></sip:<>	<sip: userc@networka;cause="302">;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)				
		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 in the last entry is set to '302'?				
	Check:	Is the "user=phone" parameter present in all History-Info header				
		URIs?				
	NOTE:	The history entries can be accumulated in "one" History-Info header				
	NOTE	or each history entry is present in one single History-Info header.				
	NOTE:	The Request line may contain a 'cause' parameter indicating the				
	Denest	redirecting reason.				
	Repeat th	nis test in reverse direction.				

Test case number	SS_cfu_(007			
Test case group		Service/CFU			
Reference	4.5.2.6/[9]				
SELECTION EXPRESSION	SE 25 AND SE 30				
Test purpose		Communication forwarding unconditional, full notification.			
	The user provided has been to URI to allows th Ensure th C, user A user C is	A and user C are in network A. The user B is in network B and is with CFU Originating user receives notification that his communication diverted = Yes and "Served user allows the presentation of forwarded originating user in diversion notification" = Yes, and "Served user e presentation of his/her URI to diverted-to user" = Yes. nat when user A calls user B, the call is forwarded unconditional to user is notified of call diversion and informed of the diverted-to number and informed of the forwarding number.			
Configuration		ption 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:				
	<sip:< th=""><th>userB@networkB>;index=1, userC@networkA;cause=302>;index=1.1</th></sip:<>	userB@networkB>;index=1, userC@networkA;cause=302>;index=1.1			
	181 Being Forwarded				
		ed-Identity: <userb@networkb></userb@networkb>			
		nfo userB@networkB>;index=1, userC@networkA;cause=302>;index=1.1			
	<sip:u< th=""><th>NVITE nfo header: userB@networkB>;index=1, userC@networkA;cause=302>;index=1.1</th></sip:u<>	NVITE nfo header: userB@networkB>;index=1, userC@networkA;cause=302>;index=1.1			
Message flow					
SIP (Network A)	÷	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFU is performed INVITE(Call-ID B-C)			
	←	181 Being Forwarded (Call-ID B-A			
	←	180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID B-A)			
	←	200 OK INVITE(Call-ID C-B) → ACK(Call-ID C-B)			
	←	200 OK INVITE(Call-ID B-A)			
	•	ACK(Call-ID A-B) → Communication Apply post test routine			
Comments	Check:	A History-Info header is received in the INVITE at the interconnection			
	Check:	interface sent to user C containing the URI identifying the served user. 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 "user=phone" parameter present in all History-Info header URIs?			
	NOTE:	The history entries can be accumulated in "one" History-Info header			
	1	or each history entry is present in one single History-Info header.			
	NOTE:	The Request line may contain a 'cause' parameter indicating the redirecting reason.			

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 uncondit	ional to user
	C user C is user determined user busy.	
Configuration		
SIP Parameter		
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network)	vork B)
	INVITE(Call-ID A-B) →	
	CFU is performed	
	← INVITE(Call-ID B-C)	
	<mark>486 Busy Here</mark> (Call-ID C-B) →	
	← ACK(Call-ID B-C)	
	486 Busy Here(Call-ID A-B)	
	ACK(Call-ID A-B) →	
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here.	
	Repeat this test in reverse direction.	

Test case number	SS_cfu_009			
Test case group	SIP-SIP/Service/CFU			
Reference	4.5.2.6/ [9]			
SELECTION EXPRESSION	SE 25			
Test purpose	Communication forwarding unconditional, unsuccessful NDUB. The user A and user C are in network A. The user B is in network B. Ensure that when user A calls user B, the call is forwarded unconditional to user C and user C is network determined user busy.			
Configuration				
SIP Parameter				
Message flow				
SIP (Network A)		Interconnection Interface		SIP (Network B)
		INVITE(Call-ID A-B)	→	
	-	CFU is performed		
	÷	INVITE(Call-ID B-C)	•	
		486 Busy Here(Call-ID C-B)	→	
	÷	ACK(Call-ID B-C)		
	÷	486 Busy Here(Call-ID A-B)		
-		ACK(Call-ID A-B)	→	
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here.			
	Repeat the	nis test in reverse direction.		

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

SS_cfu_011		
SIP-SIP/Service/CFU		
6.5/ [24]		
[Network B] SE 17 AND [Network B] SE 47 AND SE 55		
SIP-I support. CFU performed in Network B, Notification subscription options is set to presentation not allowed.		
The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number.		
Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.		
Subscription options:		
 Calling user receives notification that his call has been diverted (forwarded or deflected) = no 		
183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name]		
[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (<i>Diverted-to user</i>) Call diversion information Notification subscription options presentation not allowed Redirecting reason		
unconditional Generic notification call is diverting		
[any boundary nome]		
[any boundary name]		
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		
 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'? 		

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

Test case number	SS_cfu_014			
Test case group	SIP-SIP/Service/CFU			
Reference	6.7/[24]			
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 53			
Test purpose	SIP-I support. CFU performed in Network B, Restriction of the Redirection			
rest purpose	number.			
	number.			
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part			
	of Network B and is provided with CFU, Diverted-to user is subscribed to the			
	COLR service in Permanent mode.			
	Ensure that when user A calls user B, the call is forwarded unconditional to user			
	C, a Redirection number restriction parameter is present set to 'Presentation			
	restricted' in the encapsulated ANM contained in the 200 OK INVITE if			
	ISUP/BICC- SIP-I interworking is applicable in Network A.			
Configuration	Subscription options:			
3	 Connected user subscribed to COLR, Permanent = yes 			
SIP Parameter	200 OK			
	Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name]			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	ANM			
	Redirection number restriction			
	Presentation restricted			
	[and being demonstrated]			
Magagera flow	[any boundary name]			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)			
SIF (NELWORK A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →			
	CFU is performed			
	← INVITE(Call-ID B-C), IAM			
	180 Ringing (Call-ID C-B) →			
	← 180 Ringing (Call-ID B-A, ACM)			
	200 OK INVITE (Call-ID C-B) →			
	← ACK (Call-ID B-C)			
	← 200 OK INVITE (Call-ID B-A; ANM)			
	ACK (Call-ID A-B) →			
	Apply post test routine			
Comments	Originating user in Network A establishes a call to user in Network B. Network B			
	performs the diversion to a user in Network A.			
	Check: Is a 200 OK INVITE received at the interconnection interface?			
	Check: Is an ANM encapsulated in the 200 OK?			
	Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation			
	restricted'?			
	Repeat this test in reverse direction.			

Test case number	SS_cfu_015
Test case group	SIP-SIP/Service/CFU
Reference	6.7/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 53
Test purpose	SIP-I support. CFU performed in Network B, No restriction of the
	Redirection number.
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part
	of Network B and is provided with CFU, Diverted-to user is not subscribed to the
	COLR service.
	Ensure that when user A calls user B, the call is forwarded unconditional to user
	C, if a Redirection number restriction parameter is present it is set to
	'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE
	if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	Subscription options:
	 Connected user subscribed to COLR = no
SIP Parameter	200 OK
	Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name]
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	ANM
	Redirection number restriction
	Presentation allowed
	Or
	Redirection number restriction not present
	Redirection number restriction not present
	[any boundary name]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE(Call-ID A-B) →
	CFU is performed
	180 Ringing (Call-ID C-B) →
	 180 Ringing (Call-ID B-A, ACM)
	200 OK INVITE (Call-ID C-B) →
	← ACK (Call-ID B-C)
	← 200 OK INVITE (Call-ID B-A, ANM)
	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 group SIP-SIP/Service/CFU Reference 7.1/[24] SELECTION EXPRESSION Network B] SE 17 AND [Network B] SE 47 AND SE 55 Test purpose SIP-1 support. CFU performed in Network B, Notification of diverted-to use 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 histen 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/RICC - SIP-1 interworking is applicable in Network B. Subscription options: • SiP Parameter NUTE Content-Type: multipart/mixed;boundary=[any boundary name] - (any boundary name] Content-Type: application/isup;version=ilu-t92 Content-Type: application/isup;version=ilu-t92 Content-Disposition: signal;handling=required Address presentation restricted indicator presentation allowed Address signal Redirecting number Address presentation formation Original Redirection counter Redirection counter Redirection information Message flow Interconnection Interface INVITE(Call-ID A-B) NUTE(Call-ID A-B) NUTE(Call-ID B-C, IAM) Apply post test routine Coriginal Redirection number resent and the Address presentation restricted indicator is a user in Network A stabilistics a call to user in Network B. Network B performs the diverting number pre	Test case number	SS_cfu_016
Reference 7.1 [24] SELECTION EXPRESSION [Network B] SE 17 AND [Network B] SE 47 AND SE 55 Test purpose SIP -I support. CFU performed in Network B, Notification of diverted-to use Redirecting number 'presentation allowed'. The user A and user C are in Network B, Notification of diverted-to user diverted-to user = Release diverting number information. Ensure that when user A calls user B, Ho calls 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-1 interworking is applicable in Network B. Configuration Subscription options: • Surved user releases his/her number to diverted-to user = Release diverting number information SIP Parameter INMITE Content-Type: multipar/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/sup;version=itu-t92 Content-Disposition: signal;handling=required Address presentation netricted indicator presentation allowed Address gingal (<i>Diverting user</i>) Original called number Address gingal (<i>Diverting user</i>) Original Redirection netricted indicator presentation allowed Address gingal (<i>Diverting user</i>) Original Redirection netricted indicator presentation allowed Address gingal Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) (CHU is genformed) [any boundary name]- SIP (Network B, Network B, Network B, Network A, establishes a call to user in Network B. Network B performs the diversion to a user in Network A.		
SELECTION EXPRESSION INetwork BJ SE 17 AND [Network B] SE 47 AND SE 55 Test purpose SIP-I support. CFU performed in Network B. Notification of diverted-to user Redirecting number / presentation allowed'. The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Served user releases hisher 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. Subscription options: • Served user releases hisher number to divende-to use! = Release diverting number information? SIP Parameter INVITE Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/sup,version=itu-t92 Content-Disposition: signal;handling=required Address presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address signal (Diverting user) Original called number Redirection normation Original Redirection Reason unconditional [any boundary name]- Message flow SIP (Network A) Interconnection Interface NOTE[Call-ID A-B] NIVITE[Call-ID B-C, IAM) Apply post test routine SIP (Network B. Network B. Network A. Address presentation restricted indicator presentation allowed - CPU is performed - INVITE[Call-ID B-C, IAM) Apply post test routine SIP (Network B. Network B. Network B. Network A. Stabilishes a call to user in Network		
Test purpose SIP I support. CFU performed in Network B, Notification of diverted-to use 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 diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC > SIP-1 intervorking is applicable in Network B. Configuration Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information SIP Parameter INVITE Content-Type: nultipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-192 Content-Disposition: signal:handling=required Address gignal (Diverting user) Original called number Address presentation restricted indicator presentation allowed Address gignal (Diverting user) Original Redirection information Original Redirection information Original Redirection information Original Redirection information Original Redirection information Original Redirection ceason unknown Redirection information Original Redirection ceason UNITE Centex: Is performed (INVITE[Call-ID A-B) NVITE[Call-ID A-B, [AM] NVITE[Call-ID A-B, [AM] NVITE[Call-		
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 - SIP1 interworking is applicable in Network B. Configuration Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information if ISUP/BICC - SIP1 interworking is applicable in Network B. SIP Parameter NVITE Content-Type: multipart/mixed:boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Type: application/sup;version=itu-t92 Content-Type: application/sup;version=itu-t92 Content-Type: application allowed Address presentation restricted indicator presentation allowed Address gressentation restricted indicator presentation allowed Address gressentation restricted indicator presentation allowed Address presentati		
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-1 intervorking is applicable in Network B. Configuration Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information SIP Parameter INVITE Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal.thandling=required Address presentation restricted indicator presentation allowed Address signal (<i>Diverting user</i>) Original called number Address signal Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed Comments Originating user in Network A establishes a call to user in Network E. Network B performs the diversion to a user in Network A. Check: Is an INW necapsulated in the INVITE? Check: Is an INVITE request received at the interconnection interface? Check: Is the Redirection number present and the Address presentation restricted indicator is set to presentation allowed? Check: Is the Redirection number present and the Address presentation restricted indicator is set to presentation allowed? Check: I		
of Network B and is provided with CFU, Served user releases his/her number to diverted-to user a 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-1 intervorking is applicable in Network B. Configuration Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information SiP Parameter INVITE Content-Type: multiparl/mixed:boundary=[any boundary name] [any boundary name] Content-Type: application/sup;version=itu-192 Content-Disposition: signal (Diverting user) Original called number Address spresentation restricted indicator presentation allowed Address signal Redirecting indicator presentation information Original Redirection Reason unconditional -[any boundary name]- (any boundary name] -[any boundary name]- Message flow SIP (Network A) INVITE (Call-ID A-B) CFU is performed Criginating u		······································
of Network B and is provided with CFU, Served user releases his/her number to diverted-to user a 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-1 intervorking is applicable in Network B. Configuration Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information SiP Parameter INVITE Content-Type: multiparl/mixed:boundary=[any boundary name] [any boundary name] Content-Type: application/sup;version=itu-192 Content-Disposition: signal (Diverting user) Original called number Address spresentation restricted indicator presentation allowed Address signal Redirecting indicator presentation information Original Redirection Reason unconditional -[any boundary name]- (any boundary name] -[any boundary name]- Message flow SIP (Network A) INVITE (Call-ID A-B) CFU is performed Criginating u		The user A and user C are in Network A. The user B is in the PSTN/PLMN part
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 Redirection information if ISUP/BICC - SIP-1 interworking is applicable in Network B. Configuration Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information SIP Parameter INVITE Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-f92 Content-Type: application/isup;version=itu-f92 Content-Type: application/isup;version=itu-f92 Content-Type: application/isup;version=itu-f92 Content-Type: application/isup;version=itu-f92 Content-Disposition: signal;handling=required IAM Redirecting number Address signal (Diverting user) Original called number Address signal Redirection information Original called number Address greasention Redirection information Original Redirection interface SIP (Network A) <th></th> <th></th>		
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 presentiation allowed and Redirection information if ISUPPICC - SIP-1 interworking is applicable in Network B. 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-192 Content-Type: application/isup;version=itu-192 Content-Type: application/isup;version=itu-192 Content-Type: application/isup;version=itu-192 Content-Type: application restricted indicator presentation allowed Address gresentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirection counter Redirection counter Redirection counter Redirecting reason unconditional [any boundar		
The notification information is present in the encapsulated IAM contained in the Redirecting number presentation allowed' and Redirection information if ISUP/BICC - SIP-1 intervorking is applicable in Network B. Configuration Subscription options: • Served user releases his/her number to diverted-to user = Release divering number information SIP Parameter INVITE Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Type: application isup;version=itu-t92 Content-Type: application isup;version		Ensure that when user A calls user B, the call is forwarded unconditional to user
Redirecting number 'presentation allowed' and Redirection information if 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 signal (Diverting user) Original called number Address signal (Diverting user) Original called number Address signal Redirection information Original Redirection Reason unknown Redirection counter Redirection interface SIP (Network B) INVITE		C, user C is notified of call diversion and informed of the diverting number.
ISUP/BICC - SIP-1 interworking is applicable in Network B. Configuration Subscription options: Served user releases his/her number to diverted-to user = Release diverting number information SIP Parameter INVITE Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address signal (Diverting user) Original Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting indicator Redirection counter Redirection counter Redirection counter Redirection counter Redirection counter Redirection counter Redirection counter Redirection lnterface INVITE(Call-ID A-B) [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) [any boundary name] Comments SIP (Network B. Network B. Network B. Network B. Performed INVITE (Call-ID A-B) [any boundary name] Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection nu		The notification information is present in the encapsulated IAM contained in the
Configuration Subscription options: Served user releases his/her number to diverted-to user = Release diverting number information SIP Parameter INVITE Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address signal Redirection nestricted indicator presentation allowed Address signal Redirection interface unknown Redirection interface SIP (Network A) SIP (Network B) INVITE(Call-ID A-B) INVITE(Call-ID A-B) INVITE(Call-ID A-B) CFU is performed CFU is performed CFU is performed CFU is performed CFU is performed Comments SIP (Network B. Network B. Network B. Performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		Redirecting number 'presentation allowed' and Redirection information if
• Served user releases his/her number to diverted-to user = Release <u>idverting number information</u> SIP Parameter INVITE Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IMM Redirecting number Address presentation restricted indicator presentation allowed Address signal Cordiginal called number Address signal Redirection Reason unknown Redirection information Original Redirection Reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed (INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an IAWTE request received at the interconnection interface? Check: Is the Redirecting number resent and the Address presentation restricted indicator is set to 'presentation allowed?		ISUP/BICC - SIP-I interworking is applicable in Network B.
diverting number information SIP Parameter NVITE Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address signal Redirection information Original called number Address signal Redirection information Original Redirection counter Redirecting reason unconditional [any boundary name]- Message flow SIP (Network A) Interconnection Interface NVITE (Call-ID A-B) CFU is performed Minite guesr in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Cheeck: Is an IAM encapsulated in the INVITE? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed?	Configuration	Subscription options:
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 IMM Redirecting number Address signal (<i>Diverting user</i>) Original called number Address signal (<i>Diverting user</i>) Original Redirection restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unconditional[any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B Performs the diversion to a user in Network A. Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation allowed? Check: Is the Original called number present and the Address presentation allowed? Check: Is the Redirecting number present and the Address presentation allowed? Check: Is the Redirecting number present and the Address presentation allowed? Check: Is the Redirecting number present and the Address presentation arestricted indicator is set to 'presentation allowed? Check: Is the Redirection number present and the Address presentation arestricted indicator is set to 'presentation allowed? Check: Is the Redirection number present and the Address presentation arestricted indicator is set to 'presentation allowed? Check: Is the Redirection number present?	_	 Served user releases his/her number to diverted-to user = Release
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>Divering user</i>) Original called number Address signal (<i>Divering user</i>) Original called number Address signal (<i>Divering user</i>) Original Redirection restricted indicator presentation allowed Address signal Redirecting indicator Redirecting indicator Redirecting reason unknown Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirection ris set to 'presentation allowed'? Check: Is the Redirection inter present.		
[any boundary name] Content-Type: application/isup;version=itu-!92 Content-Disposition: signal;handling=required Medirecting number Address presentation restricted indicator presentation allowed Address signal (<i>Diverting user</i>) Original called number Address signal Redirection information Original Redirection Reason unchnown Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE[Call-ID A-B] CFU is performed Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE received at the interconnection interface? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present.	SIP Parameter	INVITE
Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required Address presentation restricted indicator presentation allowed Address signal (<i>Diverting user</i>) Original called number Address signal Redirection restricted indicator presentation restricted indicator presentation restricted indicator presentation allowed Address signal Redirection information Original Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) → CFU is performed ← INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'?		Content-Type: multipart/mixed;boundary=[any boundary name]
Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required Address presentation restricted indicator presentation allowed Address signal (<i>Diverting user</i>) Original called number Address signal Redirection restricted indicator presentation restricted indicator presentation restricted indicator presentation allowed Address signal Redirection information Original Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) → CFU is performed ← INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'?		
Content-Disposition: signal;handling=required		[any boundary name]
Imm Redirecting number Address presentation restricted indicator presentation allowed Address signal [Diverting user) Original called number Address signal [Diverting user] Original called number Address signal [Diverting user] Address signal Redirection information Original Redirection Reason unknown Redirection indicator Redirection counter Redirection counter Redirection reson unconditional [any boundary name] SIP (Network B) Message flow Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → SIP (Network B) INVITE(Call-ID B-C, IAM) Apply post test routine → Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection numb		
Redirecting number Address presentation restricted indicator presentation allowed Address signal Address signal Redirection information Original Redirection normation Original Redirection restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator 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 CFU is performed CFU is performed CFU is performed CFU is an INVITE (Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Che		Content-Disposition: signal;handling=required
Redirecting number Address presentation restricted indicator presentation allowed Address signal Address signal Redirection information Original Redirection normation Original Redirection restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator 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 CFU is performed CFU is performed CFU is performed CFU is an INVITE (Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Che		
Address presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirection counter Redirection counter Redirection counter address signal Message flow SiP (Network A) Interconnection Interface INVITE(Call-ID A-B) SiP (Network B) INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is an INVITE request received at the interconnection interface? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'?		
Presentation allowed Address signal (Diverting user) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirection information Original Redirection counter Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFU is performed CFU is performed CFU is performed Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an IAW encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection inset to 'presentation allowed'? Check: Is the Redirection number present?		
Address signal (Diverting user) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator 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 CFU is performed Moving test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is 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?		
Original called number Address presentation restricted indicator presentation allowed 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 CFU is performed Moritic Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		
Address presentation restricted indicator presentation allowed 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) [any boundary name] CFU is performed CFU is performed MVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		
presentation allowed 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 CFU is performed CFU is performed CFU is performed Credition to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection is set to 'presentation allowed'? Check: Is the Redirection interface is set to 'presentation allowed'? Check: Is the Redirection is set to 'presentation allowed'? Check: Is the Redirection is set to 'presentation allowed'? Check:		
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 an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection is set to 'presentation allowed'? Check: Is the Redirection is set to 'presentation allowed'? Check: Is the Redirection is set to 'presentation allowed'? Check: Is the Redirection is set to 'presentation allowed'? Check: Is the Redirection is set to 'presentation allowed'?		
Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network A) Interconnection Interface SIP (Network A) CFU is performed CFU is performed CFU is performed CFU is performed Coriginating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		
Original Redirection Reason unknown Redirecting indicator Redirecting counter Redirecting reason unconditional [any boundary name] Message flow SiP (Network A) Interconnection Interface SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU 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 an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		
unknown Redirecting indicator Redirection counter Redirection counter Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFU is performed (Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'?		
Redirecting indicator Redirection counter Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFU is performed CFU is performed CFU is performed Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'?		
Redirection counter Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFU is performed (NVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		
Redirecting reason unconditional [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) CFU is performed CFU is performed Moving Apply post test routine Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		
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 an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'?		
[any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) CFU is performed ← INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFU is performed CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network B Comments Originating user in Network A establishes a call to user in Network B. Network B Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFU is performed CFU is performed INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network B Comments Originating user in Network A establishes a call to user in Network B. Network B Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		[any boundary name]
SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFU is performed → K INVITE(Call-ID B-C, IAM) Apply post test routine Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'?	Message flow	
CFU 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 an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		Interconnection Interface SIP (Network B)
 INVITE (Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? 		INVITE(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 an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'?		CFU is performed
Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'?		← INVITE(Call-ID B-C, IAM)
 performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? 		
 Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? 	Comments	Originating user in Network A establishes a call to user in Network B. Network B
 Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? 		
 Check: Is the 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? 		
 restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? 		
Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'?Check: Is the Redirection number present?		
restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present?		
Check: Is the Redirection number present?		
Check: Is Redirection information present and the Redirecting reason is set to		
		Check: Is Redirection information present and the Redirecting reason is set to
'unconditional'?		
Repeat this test in reverse direction.		Repeat this test in reverse direction.

Test case number	SS_cfu_017				
Test case group	SIP-SIP/Service/CFU				
Reference	7.1/[24]				
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 55				
Test purpose	SIP-I support. CFU performed in Network B, Notification of diverted-to user				
	Redirecting number 'presentation restricted'.				
	5 1 1 1				
	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 Dedirecting number				
	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				
	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)				
Commonto	Apply post test routine				
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A.				
	Check: Is an INVITE request received at the interconnection interface?				
	Check: Is an IAM encapsulated in the INVITE?				
	Check: Is the Redirecting number present and the Address presentation				
	restricted indicator is set to 'presentation restricted'?				
	Check: Is the Original called number present and the Address presentation				
	restricted indicator is set to 'presentation restricted'?				
	Check: Is the Redirection number present?				
	Check: Is Redirection information present and the Redirecting reason is set to				
	'unconditional'?				
	Repeat this test in reverse direction.				

7.1.5.6.2 Communication Forwarding Busy (CFB)

Test case number	SS_cfb_001			
Test case group	SIP-SIP/Service/CFB			
Reference	4.5.2.6/ [9]		
SELECTION EXPRESSION	SE 26			
Test purpose	Commur	nication forwarding busy, basic	rules.	
	provided Ensure th	The user A and user C are in Network A. The user B is in network B and is provided with CFB. Ensure that when user A calls user B, the call is forwarded busy to user C. In the active call state, ensure the property of speech.		
Configuration				
SIP Parameter				
Message flow SIP (Network A)	← ← ←	Interconnection Interface INVITE(Call-ID A-B) CFB is performed INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) ACK(Call-ID B-C) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) Communication Apply post test routine	 → → → → 	SIP (Network B)
Comments	Check: Check: Check: Repeat th	CDIV busy is successful. In the active call state, ensure th Is the P-Asserted-Identity preser to Network A set to the identity on his test in reverse direction.	nt in the IN	/ITE sent from Network B

Test case number	SS_cfb_(002		
Test case group		SIP-SIP/Service/CFB		
Reference	4.5.2.6/ [4.5.2.6/ [9]		
SELECTION EXPRESSION		SE 26 AND SE 30		
Test purpose	Commu	Communication forwarding busy, no notification.		
	provided that his c Ensure th	A and user C are in Network A. with CFB, subscription option: "C communication has been diverted hat when user A calls user B, the og user is not notified.	Driginating us	ser receives notification
Configuration	Subscrip	otion options:		
	•	Originating user receives notifica diverted = No	ation that his	communication has been
SIP Parameter				
Message flow SIP (Network A)	·	Interconnection Interface INVITE(Call-ID A-B)	→	SIP (Network B)
	÷	CFB is performed INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B)	→	
	+	180 Ringing(Call-ID B-A) Apply post test routine	2	
Comments	Check:	Check: No notification regarding call forwarding in network B is received at the interconnection interface.		network B is received at the
	Repeat t	nis test in reverse direction.		

Test case number	SS_cfb_0)03		
Test case group		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 fro			
lest pulpose		ser not received.		
	Serveu u	sei not received.		
	The user	A and user C are in network A. The user B is in network B and is		
		with CFB "Originating user receives notification that his communication		
		diverted" = Yes "Served user allows the presentation of forwarded to		
		iginating user in diversion notification" = No and. "Served user allows		
	the prese	entation of his/her URI to originating user in diversion notification" = No.		
		nat when user A calls user B, the call is forwarded busy to user C, user		
	A is notified	ed of call diversion and not informed of the diverted-to number and		
		ser number.		
Configuration		otion 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	g Forwarded		
		ed-Identity: <userb@networkb></userb@networkb>		
	Privacy: i	d		
	History-In			
		<pre>userB@networkB?Privacy=history>;index=1,</pre>		
	<sip: th="" ı<=""><th>userC@networkA;cause=486?Privacy=history>;index=1.1</th></sip:>	userC@networkA;cause=486?Privacy=history>;index=1.1		
Message flow				
SIP (Network A)		Interconnection Interface SIP (Network B)		
		INVITE(Call-ID A-B) →		
	←	CFB is performed		
		INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A)		
		180 Ringing(Call-ID C-B) →		
	←	180 Ringing(Call-ID B-A)		
	•	Apply post test routine		
Comments	Check:	A 181 Being Forwarded and a History-Info header is received at the		
		interconnection interface in both entries in the History-Info header a		
		Privacy header is escaped value 'history'.		
	Check:	Is the cause parameter in the last entry set to '486'?		
	Check:	Is the "user=phone" parameter present in all History-Info header		
		URIs?		
	Check:	Is the P-Asserted-Identity header present in the 181 identifying the		
		served user?		
	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 th	his test in reverse direction.		

Test case number	SS_cfb_0	004	
Test case group		Service/CFB	
Reference	4.5.2.6/[9]		
SELECTION EXPRESSION	SE 26 AND SE 30		
Test purpose		nication forwarding busy, originating user is notified. URI from the	
		to user received.	
		A and user C are in network A. The user B is in network B and is	
		with CFB "Originating user receives notification that his communication	
		diverted" = Yes "Served user allows the presentation of forwarded to	
		iginating user in diversion notification" = Yes.	
		hat when user A calls user B, the call is forwarded busy to user C, user	
Configuration		ed of call diversion and informed of the diverted-to number.	
Configuration		otion 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		g Forwarded	
		ed-Identity: <userb@networkb></userb@networkb>	
	History-In		
		userB@networkB>;index=1,	
		userC@networkA;cause=486>;index=1.1	
Message flow			
SIP (Network A)		Interconnection Interface SIP (Network B)	
		INVITE(Call-ID A-B) →	
		CFB is performed	
	← ← 1	INVITE(Call-ID B-C)	
	~	81 Being Forwarded(Call-ID B-A) 180 Ringing(Call-ID C-B) →	
	←	180 Ringing(Call-ID B-A)	
	•	Apply post test routine	
Comments	Check:	A 181 Being Forwarded is received at interconnection interface.	
	Check:	A History-Info header is contained in the 181 with the URI of the	
		diverted-to user.	
	Check:	Is the cause parameter in the last entry set to '486'?	
	Check:	Is the "user=phone" parameter present in all History-Info header	
		URIs?	
	Check:	Is the P-Asserted-Identity header present in the 181 identifying the	
	NOTE	served user?	
1	NOTE:	The history entries can be accumulated in "one" History-Info header	
	Popost #	or each history entry is present in one single History-Info header.	
	Repeat tr	his 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		nication forwarding busy, diverted-to user does not receive the URI	
	of the se	rved user.	
		A and user C are in network C. The user B is in network B and is	
		with CFB "Served user allows the presentation of his/her URI to the to user" = No.	
		nat when user A calls user B, the call is forwarded busy "user	
		ed user busy" to user C, user C is not informed of the forwarding	
	number.	ed user busy to user C, user C is not informed of the forwarding	
Configuration		otion options:	
Comgaration	•	Served user allows the presentation of his/her URI to the diverted-to	
		user = No	
SIP Parameter	INVITE:		
	History-Ir	nfo:	
	<sip:u< th=""><th>userB@networkB<mark>?Privacy=history</mark>&Reason=SIP%3Bcause%3D486></th></sip:u<>	userB@networkB <mark>?Privacy=history</mark> &Reason=SIP%3Bcause%3D486>	
	,	ndex=1,	
	<sip:< th=""><th>userC@networkA;cause=486>;index=1.1</th></sip:<>	userC@networkA;cause=486>;index=1.1	
Message flow			
SIP (Network A)		Interconnection Interface SIP (Network B)	
		INVITE(Call-ID A-B) →	
	←		
	~	INVITE(Call-ID B-C)	
Comments	Check:	Apply post test routine A History-Info header received in the INVITE contains the URI of user	
Comments	CHECK.	B (served user) at the interconnection interface and a Privacy header	
		is escaped set to 'history'.	
	Check:	Is the cause parameter in the last entry set to '486'?	
	Check:	Is the "user=phone" parameter present in all History-Info header	
		URIs?	
	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 Request line may contain a 'cause' parameter indicating the	
		redirecting reason.	
	NOTE:	The "Reason" header in the first entry of the History-Info header sent	
		to user C is only present if the call is forwarded due to "user	
	Denerti	determined user busy" by the served user.	
	Repeat t	his test in reverse direction.	

Test case number	SS_cfb_0			
Test case group	SIP-SIP/S	SIP-SIP/Service/CFB		
Reference	4.5.2.6/ [4.5.2.6/ [9]		
SELECTION EXPRESSION	SE 26 AN	ND SE 30		
Test purpose	Commur	nication forwarding busy, diverted-to user receives the URI of the		
	served u	ser.		
	The user	A and user C are in network C. The user B is in network B and is		
		with CFB "Served user allows the presentation of his/her URI to the		
	diverted-	to user" = Yes.		
		nat when user A calls user B, the call is forwarded busy "user		
		ed user busy" to user C, user C is informed of the forwarding number.		
Configuration		otion options:		
		Served user allows the presentation of his/her URI to the diverted-to		
		<mark>user</mark> = <mark>Yes</mark>		
SIP Parameter	INVITE:			
	History-Ir			
		userB@networkB?Reason=SIP%3Bcause%3D486>;index=1,		
	<sip:< th=""><th>userC@networkA;cause=486>;index=1.1</th></sip:<>	userC@networkA;cause=486>;index=1.1		
Message flow				
SIP (Network A)		Interconnection Interface SIP (Network B)		
		INVITE(Call-ID A-B) →		
	_	CFB is performed		
	÷	INVITE(Call-ID B-C)		
-		Apply post test routine		
Comments	Check:	A History-Info header received in the INVITE contains the URI of user		
		B (served user) at the interconnection interface.		
	Check:	Is the cause parameter in the last entry set to '486'?		
	Check:	Is the "user=phone" parameter present in all History-Info header		
	NOTE	URIs?		
	NOTE:	The history entries can be accumulated in "one" History-Info header		
	NOTE	or each history entry is present in one single History-Info header.		
	NOTE:	The Request line may contain a 'cause' parameter indicating the		
	NOTE	redirecting reason.		
	NOTE:	The "Reason" header in the first entry of the History-Info header sent		
		to user C is only present if the call is forwarded due to "user		
	Derect	determined user busy" by the served user.		
	IREDeat th	nis test in reverse direction.		

Test case number	SS_cfb_(007
Test case group		Service/CFB
Reference	4.5.2.6/ [9]
SELECTION EXPRESSION		ND SE 30
Test purpose	Commur	nication forwarding busy, full notification.
	provided	A and user C are in network A. The user B is in network B and is with CFB "Originating user receives notification that his communication
		diverted" = Yes "Served user allows the presentation of forwarded to
		iginating user in diversion notification" = Yes, "diverting number is
		to the diverted-to user" = Yes.
		hat when user A calls user B, the call is forwarded busy "user
		ed user busy" to user C, user A is notified of call diversion and informed erted-to number and user C is informed of the forwarding number.
Configuration		otion options:
Comgaration		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 = <mark>Yes</mark> ,
	•	diverting number is released to the diverted-to user = Yes
SIP Parameter	INVITE:	
	History-Ir	
		userB@networkB&Reason=SIP%3Bcause%3D486>;index=1, userC@networkA;cause=486>;index=1.1
	<sip.< th=""><th>userC @networkA,cause=400>,index=1.1</th></sip.<>	userC @networkA,cause=400>,index=1.1
	181 Bein	g Forwarded
		ed-Identity: <userb@networkb></userb@networkb>
	History-Ir	
		<pre>userB@networkB>;index=1,</pre>
	<sip:< th=""><th>userC@networkA;cause=486>;index=1.1</th></sip:<>	userC@networkA;cause=486>;index=1.1
	200 OK I	
	History-Ir	iio. iserB@networkB&Reason=SIP%3B cause%3D486>;index=1,
		userC@networkA;cause=486>;index=1.1
Message flow		, , , , , , , , , , , , , , , , , , , ,
SIP (Network A)		Interconnection Interface SIP (Network B)
		INVITE(Call-ID A-B) →
		CFB is performed
	← ← /	INVITE(Call-ID B-C)
	~	I <mark>81 Being Forwarded</mark> (Call-ID B-A 180 Ringing(Call-ID C-B) →
	←	180 Ringing(Call-ID B-A)
	-	200 OK INVITE(Call-ID C-B) →
	←	ACK(Call-ID C-B)
	+	200 OK INVITE(Call-ID B-A)
		ACK(Call-ID A-B) →
		Communication
Comments	Check:	Apply post test routine A History-Info header is received in the INVITE at the interconnection
Comments	CHECK.	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 in the last entry set to '486'?
	Check:	Is the "user=phone" parameter present in all History-Info header
	NOTE	URIS?
	NOTE:	The history entries can be accumulated in "one" History-Info header
		or and history antry is present in one single Uistory lafe heads-
		or each history entry is present in one single History-Info header.
	NOTE:	The Request line may contain a 'cause' parameter indicating the
		The Request line may contain a 'cause' parameter indicating the redirecting reason.
	NOTE:	The Request line may contain a 'cause' parameter indicating the redirecting reason. The "Reason" header in the first entry of the History-Info header sent
	NOTE: NOTE:	The Request line may contain a 'cause' parameter indicating the redirecting reason.

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-ÌD A-B) →		
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here.		
	Repeat this test in reverse direction.		

Test case number	SS_cfb_	009			
Test case group	SIP-SIP/	SIP-SIP/Service/CFB			
Reference	4.5.2.6/	[9]			
SELECTION EXPRESSION	SE 26				
Test purpose	Commu	Communication forwarding busy, unsuccessful NDUB.			
	provided	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 C is network determined user busy.			
Configuration					
SIP Parameter					
Message flow	<u>.</u>				
SIP (Network A)		Interconnection Interface		SIP (Network B)	
		INVITE(Call-ID A-B)	→		
		CFB is performed			
	+	INVITE(Call-ID B-C)			
		486 Busy Here(Call-ID C-B)	→		
	+	ACK(Call-ID B-C)			
	+	486 Busy Here(Call-ID A-B)			
		ACK(Call-ID A-B)	→		
Comments	Check:	A 181 Being Forwarded is receiv	/ed at netv	vork A originating access.	
	Check:	The dialogue is terminated by re			
	Repeat t	his test in reverse direction.	0	-	

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 non trusted network.		
	The user A and user C are in network A. The network A is non trusted. The user B is in network B and is provided with CFB "Originating user receives notification that his communication has been diverted" = Yes "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No. Ensure that when user A calls user B, the call is forwarded busy to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.		
Configuration	Subscription options:		
	 Originating user receives notification that his communication has been diverted = 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 group SIP-SIP/Service/CFB Reference 6.5/[24] SELECTION EXPRESSION [Network B] SE 17 AND [Network B] SE 47 AND SE 55A 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 p of Network B and is provided with CFB, Calling user receives notification the call has been diverted (forwarded or deflected) = yes, without diverted-to use number. Ensure that when user A calls user B, the call is forwarded on busy user to u C, user A is not notified about call diversion. The notification that and Call diversion information if SIP-1 - ISUP/BICC interworking is applicable in Network B. Configuration Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = no SIP Parameter 183 Session Progress Content-Type: multipar/mixed;boundary=[any boundary name] (any boundary name] Content-Type: signal (Diverted-to user) Call diversion information Notification number Address signal (Diverted-to user) Called party's status indicator Called party's status indicator Call diversion information Notification not allowed Redirection number Addres	Test case number	SS_cfb_011		
Reference 6.5/[24] SELECTION EXPRESSION [Network B] SE 17 AND [Network B] SE 47 AND SE 55A Test purpose SIP-1 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 p of Network B and is provided with CFB, Calling user receives notification that call has been diverted (forwarded or deflected) = yes, without diverted-to use number. Ensure that when user A calls user B, the call is forwarded on busy user to u C, user A is not notified about call diversion information if SIP-1 - ISUP/BICC interworking is applicable in Network B. Configuration Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = no SIP Parameter 183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] -(any boundary name] Content-Type: multipart/mixed;boundary=[any boundary name] -(any boundary name] Content-Type: multipart/mixed;boundary=[any boundary name] Called party's status indicator Called party's status indicator Call diversion information Notification number Address signal (<i>Diverted-to user</i>) Call diversion not allowed Redirection number Address signal (<i>Diverted-to user</i>) Call diversion information Notification call is div				
SELECTION EXPRESSION [Network B] SE 17 AND [Network B] SE 47 AND SE 55.A 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 p of Network B and is provided with CFB, Calling user receives notification that call has been diverted (forwarded or deflected) = yes, without diverted-to use number. Ensure that when user A calls user B, the call is forwarded on busy user to u C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in Redirection number and Call diversion information if SIP-1 - ISUP/BICC interworking is applicable in Network B. Configuration Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = no SIP Parameter 183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-192 Content-Type: application/isup;version=itu-192 Content-Type: application number Address signal (Diverted-to user) Call diversion information Notification subscription options presentation not allowed Redirecting reason USP (Network A) Interconnection Interface SIP (Network B<				
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 p of Network B and is provided with CFB, Calling user receives notification that call has been diverted (forwarded or deflected) = yes, without diverted-to usen number. Ensure that when user A calls user B, the call is forwarded on busy user to u C, user A is not notified about call diversion information is present in the encapsulated ACM contained in Redirection number and Call diversion information if SIP-1 - ISUP/BICC interworking is applicable in Network B. Configuration Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = no SIP Parameter 183 Session Progress Content-Type: application/isup;version=itu-192 Content-Type: application/isup;version=itu-192 Content-Type: application not allowed ACM Backward call indicator Called party's status indicator Called party's status indicator Call diversion information Notification not allowed Redirection not allowed Redirection reason User Busy Generic notification Call diversion information Notification not allowed Redirection not allow				
of Network B and is provided with CFB, Calling user receives notification that call has been diverted (forwarded or deflected) = yes, without diverted-to use number. Ensure that when user A calls user B, the call is forwarded on busy user to u C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B. Configuration Subscription options: Calling user receives notification that his call has been diverted (forwarded or deflected) = no SIP Parameter 183 Session Progress Content-Type: application'isup;version=itu-t92 Content-Type: application'isup;version=itu-t92 Content-Type: application'isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator Call diversion information Redirecting reason User Busy Generic notification call is diverting [any boundary name] Message flow SIP (Network A) Interconnection Interface NPV post test routine CFB is performed Coretert is an ACM encapsuser seceived at the		SIP-I support. CFB performed in Network B, Notification subscription		
C. user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in Redirection number and Call diversion information if SIP-1 - ISUP/BICC interworking is applicable in Network B. Configuration Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = no SIP Parameter 183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 Content-Type: status indicator Called party's status indicator Call diversion information 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 INVITE(Call-ID A-B) CFB is performed <tr< th=""><th></th><th></th></tr<>				
Configuration Subscription options: Calling user receives notification that his call has been diverted (forwarded or deflected) = no SIP Parameter 183 Session Progress Content-Type: multipart/inixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (<i>Diverted-to user</i>) Call diversion information Notification subscription options presentation not allowed Redirecting reason User Busy Generic notification call is diverting [any boundary name] Message flow Interconnection Interface SIP (Network B) Interconnection Interface SIP (Network B INVITE(Call-ID B-A, ACM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Networ performed Originating user in Network A establishes a call to user in Network B. Networ performed is an ACM encapsulated in the 1837 		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		
Calling user receives notification that his call has been diverted (forwarded or deflected) = no SIP Parameter (183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (<i>Diverted-to user</i>) Call diversion information Notification subscription options presentation not allowed Redirecting reason User Busy Generic notification call is diverting [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) → CFB is performed CFB is performed Cill 32 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 Particel Check: Is a 183 Session Progress received at the interconnection interface Check: Is an ACM encapsulated in the 183?	Configuration			
Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (Diverted-to user) Call diversion information Notification subscription options presentation not allowed Redirecting reason User Busy Generic notification call is diverting [any boundary name] Message flow Interconnection Interface SIP (Network B) SIP (Network A) Interconnection Interface SIP (Network B CFB is performed INVITE(Call-ID B-C, IAM) + Comments Originating user in Network A establishes a call to user in Network B. Networ performs the diversion to a user in Network A. Check: Is a 183 Session Progress received at the interconnection interface Check: Is a ASC Session Progress received at the interconnection interface Check:		 Calling user receives notification that his call has been diverted (forwarded or deflected) = no 		
Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (Diverted-to user) Call diversion information Notification subscription options presentation not allowed Redirecting reason User Busy Generic notification call is diverting [any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) - CFB is performed (INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Networ performs the diversion to a user in Network A. Check: Is a 183 Session	SIP Parameter			
Backward call indicator Called party's status indicator no indication Redirection number Address signal (Diverted-to user) Call diversion information Notification subscription options presentation not allowed Redirecting reason User Busy Generic notification call is diverting [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network A) Interconnection Interface SIP (Network B) CFB is performed € INVITE(Call-ID B-C, IAM) € ISI Session Progress (Call-ID B-A, ACM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network A. Check: Is a 183 Session Progress		Content-Type: application/isup;version=itu-t92		
SIP (Network A) Interconnection Interface SIP (Network B INVITE(Call-ID A-B) → CFB is performed → INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network 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?		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		
Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network performs the diversion to a user in Network A. Check: Is a 183 Session Progress received at the interconnection interfac Check: Is an ACM encapsulated in the 183?	SĪP (Network A)	INVITE(Call-ID A-B) → CFB is performed ← INVITE(Call-ID B-C, IAM)		
Comments Originating user in Network A establishes a call to user in Network B. Network performs the diversion to a user in Network A. Check: Is a 183 Session Progress received at the interconnection interfac Check: Is an ACM encapsulated in the 183?		 183 Session Progress (Call-ID B-A, ACM) 		
performs the diversion to a user in Network A. Check: Is a 183 Session Progress received at the interconnection interfac Check: Is an ACM encapsulated in the 183?				
Check: Is the Redirection number present?	Comments	 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'? 		

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

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

Test case number	SS cfb 014		
Test case group	SIP-SIP/Service/CFB		
Reference	6.7/[24]		
SELECTION EXPRESSION			
	[Network B] SE 17 AND [Network B] SE 47 AND SE 55A		
Test purpose	SIP-I support. CFB performed in Network B, Restriction of the Redirection number.		
	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:		
g	 Connected user subscribed to COLR, Permanent = yes 		
SIP Parameter	200 OK		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	ANM		
	Redirection number restriction Presentation restricted		
	[any boundary name]		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B) →		
	CFB is performed		
	← INVITE(Call-ID B-C, IAM)		
	180 Ringing (Call-ID C-B) → ← 180 Ringing (Call-ID B-A, ACM)		
	 ← 180 Ringing (Call-ID B-A, ACM) 200 OK INVITE (Call-ID C-B) 		
	 ← ACK (Call-ID B-C) ← 200 OK INVITE (Call-ID B-A, ANM) 		
	ACK (Call-ID A-B) →		
	Apply post test routine		
Comments	Originating user in Network A establishes a call to user in Network B. Network B		
	performs the diversion to a user in Network A		
	Check: Is a 200 OK INVITE received at the interconnection interface?		
	Check: Is an ANM encapsulated in the 200 OK?		
	Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation		
restricted"?			
	Repeat this test in reverse direction.		

Test case number	SS_cfb_015		
Test case group	SIP-SIP/Service/CFB		
Reference	6.7/ [24]		
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 55A		
Test purpose	SIP-I support. CFB performed in Network B, No restriction of the		
rest pulpose	Redirection number.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part		
	of Network B and is provided with CFB, Diverted-to user is not subscribed to the		
	COLR service.		
	Ensure that when user A calls user B, the call is forwarded on busy user to user		
	C, if a Redirection number restriction parameter is present it is set to		
	'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE		
	if ISUP/BICC- SIP-I interworking is applicable in Network A.		
Configuration	Subscription options:		
	 Connected user subscribed to COLR = no 		
SIP Parameter	200 OK		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	ANM		
	Redirection number restriction		
	Presentation allowed		
	Or		
	Redirection number restriction not present		
	Redirection number restriction not present		
	[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)		
	180 Ringing (Call-ID C-B) →		
	← 180 Ringing (Call-ID B-A, ACM)		
	200 OK INVITE (Call-ID C-B) →		
	← ACK (Call-ID B-C)		
← 200 OK INVITE (Call-ID B-A, ANM)			
	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 an ANM encapsulated in the 200 OK?		
	Check: Is the ISUP/BICC Redirection number restriction present set to		
	'Presentation allowed' or is the parameter absent? Repeat this test in reverse direction.		

Test case number	SS_cfb_016		
Test case group	SIP-SIP/Service/CFB		
Reference	7.1/[24]		
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 55A		
Test purpose	SIP-I support. CFB performed in Network B, Notification of diverted-to user		
	Redirecting number 'presentation allowed'.		
	. .		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part		
	of Network B and is provided with CFB, Served user releases his/her number to		
	diverted-to user = Release diverting number information.		
	Ensure that when user A calls user B, the call is forwarded on busy user to user		
	C, user C is notified of call diversion and informed of the diverting number.		
	The notification information is present in the encapsulated IAM contained in the		
	Redirecting number 'presentation allowed' and Redirection information if		
	ISUP/BICC - SIP-I interworking is applicable in Network B.		
Configuration	Subscription options:		
	 Served user releases his/her number to diverted-to user = Release diverting 		
	number information		
SIP Parameter	INVITE		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	IAM		
	Redirecting number		
	Address presentation restricted indicator		
	presentation allowed		
	Address signal (<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		
	← INVITE(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 an INVITE request received at the interconnection interface?		
	Check: Is an IAM encapsulated in the INVITE?		
	Check: Is the Redirecting number present and the Address presentation		
	restricted indicator is set to 'presentation allowed'?		
	Check: Is the Original called number present 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.		
L			

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

7.1.5.6.3 Communication Forwarding No Reply (CFNR)

Test case number	SS cfnr	001			
Test case group	SIP-SIP/Service/CFNR				
Reference	4.5.2.6/ [9]			
SELECTION EXPRESSION	SE 27				
Test purpose	Commu	nication forwarding no reply, ba	sic rules.		
		5 1, 9, 1			
	The user	A and user C are in Network A. T	he user B i	s in network B and is	
	provided	with CFNR.			
	Ensure t	hat when user A calls user B, the c	call is forwa	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) →			
	← ←				
	T	200 OK INVITE(Call-ID B-A)	→		
		ACK(Call-ID A-B) Communication	7		
		Apply post test routine			
Comments	Check:	CDIV no reply is successful.			
Commenta	Check:		e nronertv (ofspeech	
	Check: In the active call state, ensure the property of speech. Check: Is the P-Asserted-Identity present in the INVITE sent from Network				
	5	to Network A set to the identity of			
	Repeat t	his test in reverse direction.	o origine		

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	
lest purpose	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	
	P-Asserted-Identity: <userb@networkb></userb@networkb>	
	Privacy: id	
	History-Info:	
	<sip:userb@networkb?privacy=history>;index=1,</sip:userb@networkb?privacy=history>	
	<sip: userc@networka;cause="408?Privacy=history">;index=1.1</sip:>	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
	INVITE(Call-ID A-B) →	
	← 180 Ringing(Call-ID B-A)	
	CFB is performed	
	← INVITE(Call-ID B-C)	
	 181 Being Forwarded(Call-ID B-A) 180 Ringing(Call-ID C-B) → 	
	 180 Ringing(Call-ID B-A) Apply post test routine 	
Comments	Check: A 181 Being Forwarded and a History-Info header is received at the	
Comments	interconnection interface in both entries in the History-Info header a	
	Privacy header is escaped value 'history'.	
	Check: Is the cause parameter in the last entry set to '408'?	
	Check: Is the "user=phone" parameter present in all History-Info header	
	URIs?	
	Check: Is the P-Asserted-Identity header present in the 181 identifying the	
	served user?	
	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.	
	Depend 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 actilization 		
SIP Parameter	in diversion notification = Yes 181 Being Forwarded		
SIP Parameter	P-Asserted-Identity: <userb@networkb></userb@networkb>		
	History-Info:		
	<sip:userb@networkb>;index=1,</sip:userb@networkb>		
	<sip: userc@networka;cause="408">;index=1.1</sip:>		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B) →		
	 180 Ringing(Call-ID B-A) 		
	CFB is performed		
	← INVITE(Call-ID B-C)		
	← 181 Being Forwarded(Call-ID B-A)		
	180 Ringing(Call-ID C-B) →		
	← 180 Ringing(Call-ID B-A)		
0	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		
	Check: A History-Info header is contained in the 181 with the URI of the		
	diverted to upor		
	diverted-to user.		
	Check: Is the cause parameter in the last entry is set to '408'?		
	Check: Is the cause parameter in the last entry is set to '408'?Check: Is the "user=phone" parameter present in all History-Info header URIs?		
	 Check: Is the cause parameter in the last entry is set to '408'? Check: Is the "user=phone" parameter present in all History-Info header URIs? Check: Is the P-Asserted-Identity header present in the 181 identifying the 		
	 Check: Is the cause parameter in the last entry is set to '408'? Check: Is the "user=phone" parameter present in all History-Info header URIs? Check: Is the P-Asserted-Identity header present in the 181 identifying the served user? 		
	 Check: Is the cause parameter in the last entry is set to '408'? Check: Is the "user=phone" parameter present in all History-Info header URIs? Check: Is the P-Asserted-Identity header present in the 181 identifying the served user? NOTE: The history entries can be accumulated in "one" History-Info header 		
	 Check: Is the cause parameter in the last entry is set to '408'? Check: Is the "user=phone" parameter present in all History-Info header URIs? Check: Is the P-Asserted-Identity header present in the 181 identifying the served user? 		

Test sees number	CC of an	005	
Test case number	SS_cfnr_005		
Test case group	SIP-SIP/Service/CFNR		
Reference	4.5.2.6/ [
SELECTION EXPRESSION	SE 27 AN	ND SE 30	
Test purpose		nication forwarding no reply, diverted-to user does not receive the ne served user.	
	provided to user" = Ensure th	hat when user A calls user B, the call is forwarded no reply to user C,	
		not informed of the forwarding number.	
Configuration		otion options:	
	 Serv 	ed user allows the presentation of his/her URI to the diverted-to user =	
	No		
SIP Parameter	INVITE		
	History-Ir		
	<mark><sip:ເ< mark=""></sip:ເ<></mark>	userB@networkB?Privacy=history>;index=1,	
	<sip:< th=""><th>userC@network1;cause=408>;index=1.1</th></sip:<>	userC@network1;cause=408>;index=1.1	
Message flow			
SIP (Network A)		Interconnection Interface SIP (Network B)	
		INVITE(Call-ID A-B) →	
	←	180 Ringing(Call-ID B-A)	
		CFB is performed	
	←	INVITE(Call-ID B-C)	
		Apply post test routine	
Comments	Check:	A History-Info header 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 in the last entry is set to '408'?	
	Check:	Is the "user=phone" parameter present in all History-Info header URIs?	
	NOTE:	The history entries can be accumulated in "one" History-Info header	
	1	or each history entry is present in one single History-Info header.	
	NOTE:	The Request line may contain a 'cause' parameter indicating the redirecting reason.	
	Repeat th	his test in reverse direction.	
	1		

Test case number	SS_cfnr	006	
Test case group		 Service/CFNR	
Reference	4.5.2.6/ [
SELECTION EXPRESSION		ND SE 30	
Test purpose	-	nication forwarding no reply, diverted-to user receives the URI of	
		rted-to user.	
	The user A and user C are in network A. The user B is in network B and is		
	provided to user" =	with "Served user allows the presentation of his/her URI to the diverted- = Yes.	
	Ensure t	hat when user A calls user B, the call is forwarded no reply to user C,	
	user C is	informed of the forwarding number.	
Configuration	Subscri	otion options:	
		ed user allows the presentation of his/her URI to the diverted-to user =	
	Yes		
SIP Parameter	INVITE		
	History-I		
		userB@networkB>;index=1,	
	<sip:< th=""><th>userC@network1;cause=408>;index=1.1</th></sip:<>	userC@network1;cause=408>;index=1.1	
Message flow		lutere environtion luterform	
SIP (Network A)		Interconnection Interface SIP (Network B) INVITF(Call-ID A-B) →	
	←	INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A)	
	C	CFB is performed	
	÷	INVITE(Call-ID B-C)	
	•	Apply post test routine	
Comments	Check:	A History-Info header is received in the INVITE contains the URI of	
		user B (served user) at the interconnection interface.	
	Check:	Is the 'cause' parameter present in the Request line sent to user C	
		(diverted-to user) set to '408'?	
	Check:	Is the "user=phone" parameter present in all History-Info header	
		URIs?	
	Check:	Is the P-Asserted-Identity header present in the 181 identifying the	
		served user?	
	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 Request line may contain a 'cause' parameter indicating the	
		rodirocting roacon	
	Decest	redirecting reason. his test in reverse direction.	

Test case number	SS_cfnr_0	007	
Test case group	SIP-SIP/Service/CFNR		
Reference	4.5.2.6/ [9]		
SELECTION EXPRESSION	SE 27 AN		
Test purpose		ication forwarding no reply, full notification.	
		A and user C are in network A. The user B is in network B and is with CFNR "Originating user receives notification that his	
		ation has been diverted" = Yes, "Served user allows the presentation	
		ed to URI to originating user in diversion notification" = Yes, "diverting	
		released to the diverted-to user" = Yes.	
		at when user A calls user B, the call is forwarded no reply to user C,	
		notified of call diversion and informed of the diverted-to number and	
		nformed of the forwarding number.	
Configuration	Subscript	tion options:	
		nating user receives notification that his communication has been	
		ed = Yes	
		ed user allows the presentation of forwarded to URI to originating user	
	in div	ersion notification = Yes	
		ing number is released to the diverted-to user = Yes	
SIP Parameter	INVITE:		
	History-Inf	fo:	
	<sip:us< th=""><th>serB@networkB>;index=1,</th></sip:us<>	serB@networkB>;index=1,	
	<sip: th="" u<=""><th><pre>userC@networkA;cause=486>;index=1.1</pre></th></sip:>	<pre>userC@networkA;cause=486>;index=1.1</pre>	
		Forwarded	
		d-Identity: <userb@networkb></userb@networkb>	
	History-Inf		
		serB@network>;index=1,	
	<sip: th="" u<=""><th><pre>userC@networkA;cause=408>;index=1.1</pre></th></sip:>	<pre>userC@networkA;cause=408>;index=1.1</pre>	
	200 OK IN History-Inf		
	<sip:us< th=""><th>serB@networkB>;index=1,</th></sip:us<>	serB@networkB>;index=1,	
Message flow	<sip:us< th=""><th></th></sip:us<>		
Message flow SIP (Network A)	<sip:us< th=""><th>serB@networkB>;index=1, iserC@networkA;cause=408>;index=1.1</th></sip:us<>	serB@networkB>;index=1, iserC@networkA;cause=408>;index=1.1	
Message flow SIP (Network A)	<sip:us< th=""><th>serB@networkB>;index=1, IserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B)</th></sip:us<>	serB@networkB>;index=1, IserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B)	
	<sip:us< th=""><th>serB@networkB>;index=1, IserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →</th></sip:us<>	serB@networkB>;index=1, IserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →	
	<sip:us <sip: th="" u<=""><th>serB@networkB>;index=1, IserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A)</th></sip:></sip:us 	serB@networkB>;index=1, IserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A)	
	<sip:us <sip: th="" u<=""><th>serB@networkB>;index=1, IserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed</th></sip:></sip:us 	serB@networkB>;index=1, IserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed	
	<sip:us <sip: u<br="">←</sip:></sip:us 	serB@networkB>;index=1, iserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C)	
	<sip:us <sip: u<br="">←</sip:></sip:us 	serB@networkB>;index=1, IserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed	
	<sip:us <sip: u<br="">←</sip:></sip:us 	serB@networkB>;index=1, iserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A	
	<sip:us <sip: u<br="">←</sip:></sip:us 	serB@networkB>;index=1, iserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) →	
	<sip:us <sip: u<br="">< < <</sip:></sip:us 	serB@networkB>;index=1, iserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B)	
	<sip:us <sip: u<br="">< < • • 11 •</sip:></sip:us 	serB@networkB>;index=1, iserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID C-B) 200 OK INVITE(Call-ID B-A)	
	<sip:us <sip: u<br="">< < <</sip:></sip:us 	serB@networkB>;index=1, iserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) \rightarrow 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) \rightarrow 180 Ringing(Call-ID C-B) \rightarrow 180 Ringing(Call-ID C-B) \rightarrow 180 Ringing(Call-ID C-B) \rightarrow 200 OK INVITE(Call-ID C-B) ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) \rightarrow	
SIP (Network A)	<sip:us <sip: u<br="">< + + + 1 + + + + + +</sip:></sip:us 	serB@networkB>;index=1, iserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID C-B) ACK(Call-ID A-B) → Apply post test routine	
	<sip:us <sip: u<br="">< < <</sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 200 OK INVITE(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) →	
SIP (Network A)	<sip:us <sip: u<br=""><sip: u<br="">< < < < < < < < < < < < < <</sip:></sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) →	
SIP (Network A)	<sip:us <sip: u<br="">< + + + 1 + + + + + +</sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID B-A) ACK(Call-ID B-A) ACK(Call-ID B-A) ACK(Call-ID B-A) ACK(Call-ID B-A) ACK(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(C	
SIP (Network A)	<sip:us <sip: u<br=""><sip: u<br="">< < < < < < < < < < < < < <</sip:></sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Cal	
SIP (Network A)	<sip:us <sip: u<br=""><sip: th="" u<=""><th>serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-</th></sip:></sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-	
SIP (Network A)	<sip:us <sip: u<br="">< < < < < < 1</sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Cal	
SIP (Network A)	<sip:us <sip: u<br=""><sip: th="" u<=""><th>serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → Iterface sent to user C containing the URI identifying the served user. A History-Info header is received in the 181 Being Forwarded at the interface sent to user C containing the URI identifying the served user. 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. Is the cause parameter in the last entry is set to '408'? Is the "user=phone" parameter present in all History-Info header</th></sip:></sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → Iterface sent to user C containing the URI identifying the served user. A History-Info header is received in the 181 Being Forwarded at the interface sent to user C containing the URI identifying the served user. 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. Is the cause parameter in the last entry is set to '408'? Is the "user=phone" parameter present in all History-Info header	
SIP (Network A)	<sip:us <sip: u<br="">< < < < < < 1</sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → Iterface sent to user C containing the URI identifying the served user. A History-Info header is received in the 181 Being Forwarded at the interface sent to user C containing the URI identifying the served user. 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. Is the cause parameter in the last entry is set to '408'? Is the "user=phone" parameter present in all History-Info header URIs?	
SIP (Network A)	<sip:us <sip: u<br="">< < < < < < 1</sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → Interface sent to user C containing the URI identifying the served user. A History-Info header is received in the 181 Being Forwarded at the interface sent to user C containing the URI identifying the served user. 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. Is the cause parameter in the last entry is set to '408'? Is the "user=phone" parameter present in all History-Info header URIs? Is the P-Asserted-Identity header present in the 181 identifying the	
SIP (Network A)	<sip:us <sip: u<br=""><sip: u<br="">< < < < < < 1</sip:></sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → Interface sent to user C containing the URI identifying the served user. A History-Info header is received in the 181 Being Forwarded at the interface sent to user C containing the URI identifying the served user. 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. Is the cause parameter in the last entry is set to '408'? Is the "user=phone" parameter present in all History-Info header URIs? Is the P-Asserted-Identity header present in the 181 identifying the served user?	
SIP (Network A)	<sip:us <sip: u<br="">< < < < < < 1</sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID A-B) → Itstory-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. 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. Is the cause parameter in the last entry is set to '408'? Is the "user=phone" parameter present in all History-Info header URIs? Is the P-Asserted-Identity header present in the 181 identifying the served user? The history entries can be accumulated in "one" History-Info header	
SIP (Network A)	<sip:us <sip: u<br=""><sip: u<br="">< < < < < 1</sip:></sip:></sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → Ition Street routine A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. 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. Is the cause parameter in the last entry is set to '408'? Is the "user=phone" parameter present in all History-Info header URIs? Is the P-Asserted-Identity header present in the 181 identifying the served user? 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)	<sip:us <sip: u<br=""><sip: u<br="">< < < < < < 1</sip:></sip:></sip:us 	serB@networkB>;index=1, IserC@networkA;cause=408>;index=1.1 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 200 OK INVITE(Call-ID B-A) 200 OK INVITE(Call-ID B-A) ACK(Call-ID C-B) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → Apply post test routine A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. 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. Is the cause parameter in the last entry is set to '408'? Is the "user=phone" parameter present in all History-Info header URIs? Is the P-Asserted-Identity header present in the 181 identifying the served user? The history entries can be accumulated in "one" History-Info header. The Request line may contain a 'cause' parameter indicating the	
SIP (Network A)	<sip:us <sip:us <sip:u < < < < < < 1 < < < < < < < < < < < <</sip:u </sip:us </sip:us 	serB@networkB>;index=1, Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing(Call-ID B-A) CFB is performed INVITE(Call-ID B-C) 81 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → 180 Ringing(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID C-B) → ACK(Call-ID B-A) 200 OK INVITE(Call-ID B-A) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → ACK(Call-ID A-B) → Itstory-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. 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. Is the cause parameter in the last entry is set to '408'? Is the "user=phone" parameter present in all History-Info header URIs? Is the P-Asserted-Identity header present in the 181 identifying the served user? 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 number	SS_cfnr_008
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/ [9]
SELECTION EXPRESSION	SE 27
Test purpose	Communication forwarding no reply, unsuccessful UDUB.
	The user A and user C are in network A. The user B is in network B and is provided with CFNR. Ensure that when user A calls user B, the call is forwarded no reply to user C and user C is user determined user busy.
Configuration	
SIP Parameter	
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →
	 180 Ringing(Call-ID B-A)
	CFB is performed ← INVITE(Call-ID B-C) 486 Busy Here(Call-ID C-B) →
	← ACK(Call-ID B-C)
	
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction.

Test case number	SS_cfnr_	009		
Test case group		SIP-SIP/Service/CFNR		
Reference	4.5.2.6/ [9	9]		
SELECTION EXPRESSION	SE 27			
Test purpose	Commur	nication forwarding no reply, un	successfu	II NDUB.
	provided Ensure th	A and user C are in network A. Th with CFNR. hat when user A calls user B, the c 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-ID B-C)	2	
	÷	486 Busy Here(Call-ID A-B)		
		ACK(Call-ID A-B)	→	
Comments	Check:	The dialogue is terminated by re	ceiving a 4	86 Busy Here.
	Repeat th	his test in reverse direction.	Ū.	-

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

Test case number	SS_cfnr_011
Test case group	SIP-SIP/Service/CFNR
Reference	6.5/ [24]
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 55B
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:
Connguration	 Calling user receives notification that his call has been diverted (forwarded or deflected) = no
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required
	CPG Event indicator
	Alerting or Progress
	Redirection number
	Address signal (Diverted-to user)
	Call diversion information
	Notification subscription options
	presentation not allowed
	Redirecting reason
	No reply Generic notification
	call is diverting
	[any boundary name]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing (Call-ID B-A, ACM) CFNR is performed INVITE(Call-ID B-C, IAM)
	 183 Session Progress (Call-ID B-A, CPG)
	Apply post test routine
Comments	Originating user in Network A establishes a call to user in Network B. Network B
	performs the diversion to a user in Network A.
	Check: Is a 183 Session Progress received at the interconnection interface?
	Check: Is a CPG encapsulated in the 183?
	Check: Is the Called party's status indicator set to 'no indication'?
	Check: Is the Redirection number present?
	Check: Is Notification subscription options indicator set to 'presentation not allowed'?
	Check: Is the Redirecting reason set to 'No reply'? Repeat this test in reverse direction.

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

Test case number	SS_cfnr_013
Test case group	SIP-SIP/Service/CFNR
Reference	6.5/ [24]
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 55B
Test purpose	SIP-I support. CFNR performed in Network B, Notification subscription options is set to presentation allowed with redirection number.
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number. Ensure that when user A calls user B, the call is forwarded on no reply to user C,
	user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated CPG contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options:
	 Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name]
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required
	CPG Event indicator
	Alerting or Progress
	Redirection number
	Address signal (Diverted-to user)
	Call diversion information
	Notification subscription options
	presentation allowed with redirection number Redirecting reason
	No reply
	Generic notification
	call is diverting
Message flow	[any boundary name]
SIP (Network A)	 Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing (Call-ID B-A, ACM) CFNR is performed INVITE(Call-ID B-C, IAM)
	 183 Session Progress (Call-ID B-A, CPG) Apply post test routine
Comments	Originating user in Network A establishes a call to user in Network B. Network B
	performs the diversion to a user in Network A.
	Check: 183 Session Progress is received at the interconnection interface. Check: Is a CPG encapsulated in the 183?
	Check: Is a CPG encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'?
	Check: Is the Redirection number present?
	Check: Is Notification subscription options indicator is set to 'presentation allowed with redirection number'?
	Check: Is the Redirecting reason set to 'No reply'? Repeat this test in reverse direction.

Test case number	SS_cfnr_014	
Test case group	SIP-SIP/Service/CFNR	
Reference	6.7/[24]	
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 53B	
Test purpose	SIP-I support. CFNR performed in Network B, Restriction of the Redire	
	number.	
	The wave A and wave O are in Naturaly A. The wave D is in the DOTN/DUMN and	
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part	
	of Network B and is provided with 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:	
Comgaration	 Connected user subscribed to COLR, Permanent = yes 	
SIP Parameter	200 OK	
	Content-Type: multipart/mixed;boundary=[any boundary name]	
	······································	
	[any boundary name]	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	ANM	
	Redirection number restriction	
	Presentation restricted	
	[any boundary name]	
Message flow	Interconnection Interface CID (Network D)	
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)	
	180 Ringing (Call-ID C-B) →	
	← 180 Ringing (Call-ID B-A, ACM)	
	200 OK INVITE (Call-ID C-B) →	
	← ACK (Call-ID B-C)	
	← 200 OK INVITE (Call-ID B-A; ANM)	
	$\frac{200 \text{ CK INVITE}}{\text{ACK (Call-ID A-B)}} \rightarrow $	
	Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B	
	performs the diversion to a user in Network A.	
	Check: Is a 200 OK INVITE received at the interconnection interface?	
	Check: Is an ANM encapsulated in the 200 OK?	
	Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation	
	restricted'?	
	Repeat this test in reverse direction.	

Test case number	SS_cfnr_015				
Test case group	SIP-SIP/Service/CFNR				
Reference	6.7/ [24]				
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 55B				
Test purpose	SIP-I support. CFNR performed in Network B, No restriction of the				
	Redirection number.				
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part				
	of Network B and is provided with CFNR, Diverted-to user is not subscribed to				
	the COLR service.				
	Ensure that when user A calls user B, the call is forwarded on no reply to user C,				
	if a Redirection number restriction parameter is present it is set to 'Presentation				
	allowed' in the encapsulated ANM contained in the 200 OK INVITE if				
	ISUP/BICC- SIP-I interworking is applicable in Network A.				
Configuration	Subscription options:				
	 Connected user subscribed to COLR = no 				
SIP Parameter	200 OK				
	Content-Type: multipart/mixed;boundary=[any boundary name]				
	[any boundary name]				
	Content-Type: application/isup;version=itu-t92				
	Content-Disposition: signal;handling=required				
	ANM				
	Redirection number restriction				
	Presentation allowed				
	or				
	Redirection number restriction not present				
	[any boundary name]				
Message flow					
SIP (Network A)	Interconnection Interface SIP (Network B)				
	INVITE(Call-ID A-B)				
	← 180 Ringing (Call-ID B-A, ACM)				
	CFNR is performed				
	← INVITE(Call-ID B-C, IAM)				
	180 Ringing (Call-ID C-B) →				
	← 180 Ringing (Call-ID B-A, ACM)				
	200 OK INVITE (Call-ID C-B) →				
	← ACK (Call-ID B-C)				
	← 200 OK INVITE (Call-ID B-A, ANM)				
	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.				

Fest case group SIP-SerVice/CFNR Seference 7./1 [24] SELECTION EXPRESSION INetwork B] SE 17 AND [Network B] SE 47 AND SE 55B SIP-I support. CFNR performed 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 information of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection notomation if ISUPRICC - SIP-1 intervorking is applicable in Network B. Subscription options: • Subscription options: • Subscription options: • Sarved user releases his/her number to diverted-to user Release diverting number information Redirecting number / second Redirection information if ISUPRICC - SIP-1 intervorking is applicable in Network B. Subscription options: • Sarved user releases his/her number to diverted-to user Reliase diverting number / second number / second Redirection network B. Subscription options: • Sarved user releases pliciton/isupversion=itu-192 Content-Type: multipart/mixed;boundary=[any boundary name] (any boundary name] -(any boundary name] Original called number Address gresentation restricted indicator presentation allowed Address signal Not reply • Too Redirecting number <t< th=""><th>Test case number</th><th>SS_cfnr_016</th></t<>	Test case number	SS_cfnr_016				
ELECTION EXPRESSION [Network B] SE 17 AND [Network B] SE 17 AND SE 55E Fest purpose SIP-1 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 information is present in the encapsulated IAM contained in the Redirecting number information is present in the encapsulated IAM contained in the Redirecting number information and Redirection information if SUP-PRICC. SIP-I intervorking is applicable in Network B. Configuration Subscription options: 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] - (any boundary name] Content-Type: application/isup:version=itu-192 Content-Type: application restricted indicator presentation allowed Address signal (Dwerting user) Original called number Address presentation restricted indicator presentation allowed Address signal (Dwerting user) • Original called number • Address gresson No reply - (any boundary name] • <th>Test case group</th> <th colspan="4"></th>	Test case group					
Fest purpose SIP-1 support. CFNR performed in Network B, Notification of diverted-to user Redirecting number ipresentation 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 a Calls user B, the call is forwarded on no reply to user C, user C is notified of call diversion information is present in the encapsulated IAM contained in the Redirecting number. The notification information is present in the encapsulated IAM contained in the Redirecting number. The notification information is present in the encapsulated IAM contained in the Redirecting number. SIP-FIRENOVK B. Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number. Information SIP Parameter INVITE Content-Type: enultipart/mixed; boundary=[any boundary name] - (any boundary name] - (any boundary name] Content-Type: spresentation restricted indicator - (any boundary name] - (any boundary name]	Reference	7.1/[24]				
Fest purpose SIP-1 support. CFNR performed in Network B, Notification of diverted-to user Redirecting number ipresentation 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 a Calls user B, the call is forwarded on no reply to user C, user C is notified of call diversion information is present in the encapsulated IAM contained in the Redirecting number. The notification information is present in the encapsulated IAM contained in the Redirecting number. The notification information is present in the encapsulated IAM contained in the Redirecting number. SIP-FIRENOVK B. Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number. Information SIP Parameter INVITE Content-Type: enultipart/mixed; boundary=[any boundary name] - (any boundary name] - (any boundary name] Content-Type: spresentation restricted indicator - (any boundary name] - (any boundary name]	SELECTION EXPRESSION					
of Network B and is provided with CFNR, Served user releases his/her number to divered-to user R elaeses divering number information. Ensure that when user A calls user B, the call is forwarded on or 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-1 intervorking is applicable in Network B. Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information SUBScription options: • Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal,handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address signal Redirection information Original Redirection Reason unknown Redirection network P. SIP (Network A) Interconnection Interface SIP (Network A) Interconnection Interface NVITE[Call-ID A-B] • 180 Ringing (Call-ID A-B) • 180 Ringing (Call-ID B-A, ACM) CFNR is performed Crinet as user in Network A. Network B. Network B. No reply [any boundary name]- Content: Is an IAW encapsulated in the INVTEF? Check: Is an IAW encapsulated in the INVTEF? Check: Is an IAW encapsulated in the INVTEF? Check: Is the Original called number resent and the Address presentation restricted indicator is set to present and the Address presentation restricted indicator is set to present and the Address presentation restricted indicator is set to present and the Address presentation restricted indicator is set to present and the Address presentation restricted indicator is set to present and the Address presentation restricted indicator is set to present and the Address presentation restricted indicator is set to present and the Address present	Test purpose	SIP-I support. CFNR performed in Network B, Notification of diverted-to				
User C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC - SIP-I intervorking is applicable in Network B. Configuration Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information SIP Parameter INVITE Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 Content-Type: application restricted indicator presentation restricted indicator presentation allowed Address presentation restricted indicator presentation allowed Address signal Redirecting number Redirection information Wessage flow Interconnection Interface Normed SIP (Network A) VITE(Call-ID A-B) No reply [any boundary name] Versage flow Interconnection Interface No reply Comments Original caller on unknown Redirecting indicator Redirecting reason No reply Comments Originating user in Network A Address presentation Redirecting call-ID B-A, ACM) CFIRR is performed Check: Is an INVITE (Call-ID B-C, IAM) Apply post test routine Cordinating user in Network A checks presentation restricted indicator is set to present and the Address presentation restricted indicator is set to present and the Address presentation restricted indicator is set to present restricted indicator		of Network B and is provided with CFNR, Served user releases his/her number to diverted-to user = Release diverting number information.				
Served user releases his/her number to diverted-to user = Release divering number information INVITE Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address signal (Diverting user) Original called number Address signal Redirection restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting indicator Redirecting indicator Redirecting reason No reply [any boundary name] Wessage flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE (Call-ID B-A, ACM) CFNR is performed for signating user in Network A call to user in Network B. Network B performs the diversion to a user in Network B. Network B. Performs the diversion to a user in Network A. Check: Is an IAM encapsulated in the INVITE? Check: Is an IAM encapsulated in the INVITE? Check: Is an IAM encapsulated in the NVITE? Check: Is the Redirection number present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present information Redirection number present? Check: Is the Redirection number present?		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				
INVITE SiP Parameter NUTTE 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 spresentation restricted indicator presentation allowed Address signal Redirecting number Address signal Redirection information Original called number Address signal Redirection information Original Redirection Reason unknown Redirection counter Redirection indicator Redirection indicator Redirection counter Redirection seson No reply [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID B-C, IAM) CFNR is performed INVITE (Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A sestablishes a	Configuration	Subscription options:				
SIP Parameter INVITE Content-Type: multipart/mixed;boundary=[any boundary name][any boundary name] Content-Type: application/sup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason No reply[any boundary name]- Vessage flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an IAM Encapsulated in the INVITE? Check: Is an IAM encapsulated in the INVITE? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirection number present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address presentation restricted indicator is to present and the Address pre	-					
Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address signal Redirection nestricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason No reply [any boundary name] Wessage flow SIP (Network A) Interconnection Interface SIP (Network A) INVITE (Call-ID A-B) Apply post test routine Check: Is an INVITE request received at the interconnection interface? Check: Is an INWITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is an IAM encapsulate		number information				
Content-Type: application/isup;version=itu-192 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address signal Redirection information Original Redirection Reason unknown Redirection information Original Redirection Reason unknown Redirection counter Redirection counter Redirection counter Redirection interface SIP (Network B) INVITE(Call-ID A-B) → (E) 180 Ringing (Call-ID B-A, ACM) CFNR is performed (E) INVITE(Call-ID A-B, D,	SIP Parameter					
Redirecting number Address presentation allowed Address signal (Diverting user) Original called number Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting indicator Redirecting indicator Redirecting reason No reply [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network A) CFNR is performed € 180 Ringing (Call-ID A-B) • 180 Ringing (Call-ID B-C, IAM) Apply post test routine Check: Is an INVITE (Call-ID B-C, IAM) Apply post test routine Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to presentation allowed? Check: Is the Redirecting number present? Check: Is the Redirecting number present? Check:		Content-Type: application/isup;version=itu-t92				
Address presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address signal (Diverting user) Original called number Address presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirection counter Redirection counter SIP (Network A) SIP (Network B) SIP (Network B) INVITE (Call-ID A-B) INVITE (Call-ID B-A, ACM) CFINR is performed (INVITE (Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is the Redirection number present and the Redirecting reason is set to 'No reply'?		IAM				
Address presentation restricted indicator presentation allowed Address signal (Diverting user) Original called number Address signal (Diverting user) Original called number Address presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirection counter Redirection counter SIP (Network A) SIP (Network B) SIP (Network B) INVITE (Call-ID A-B) INVITE (Call-ID B-A, ACM) CFINR is performed (INVITE (Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is the Redirection number present and the Redirecting reason is set to 'No reply'?						
Address signal (Diverting user) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirection counter Redirection counter Redirection reason No reply [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) 180 Ringing (Call-ID B-A, ACM) CFNR is performed INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an IAVITE request received at the interconnection interface? Check: Is an IAVITE request received at the interconnection interface? Check: Is an IAVITE request received at the interconnection interface? Check: Is an IAVITE request received at the interconnection interface? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present? Check: Is the Redirection number present? Check: Is the Redirection number present? Check: Is the Redirection information p						
Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting indicator Redirecting reason No reply [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) ← 180 Ringing (Call-ID B-A, ACM) CFNR is performed ← INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an IAM encapsulated in the INVITE? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is t						
Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason No reply [any boundary name] Wessage flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE (Call-ID A-B) ✓ 180 Ringing (Call-ID B-A, ACM) CFNR is performed ✓ Mortific Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection information present? Check: Is the Redirection number						
presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason No reply [any boundary name] Wessage flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) ← 180 Ringing (Call-ID B-A, ACM) CFNR is performed ← INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is an IAM encapsulated in the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is the Redirection number present? Check:						
Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason No reply [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) ← 180 Ringing (Call-ID B-A, ACM) CFNR is performed ← INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection information present and the Redirecting reason is set to 'No reply'?		Address presentation restricted indicator				
Redirection information Original Redirection Reason unknown Redirecting indicator Redirecting reason No reply [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) ← 180 Ringing (Call-ID B-A, ACM) CFNR is performed ← INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection information present? Check: Is the Redirection information present and the Redirecting reason is set to 'No reply'?						
Original Redirection Reason unknown Redirecting indicator Redirecting cason No reply [any boundary name] Wessage flow SIP (Network A) Interconnection Interface 180 Ringing (Call-ID A-B) + € 180 Ringing (Call-ID B-A, ACM) CFNR is performed € € 180 Ringing (Call-ID B-A, ACM) CFNR is performed € Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an 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 Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present? Check: Is the Redirection number present? Check: Is the Redirection number present?						
unknown Redirecting indicator Redirecting indicator Redirecting reason No reply [any boundary name] Vessage flow Interconnection Interface SIP (Network B) SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A, ACM) CFNR is performed ← ← INVITE(Call-ID B-C, IAM) Apply post test routine Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is the Redirection						
Redirecting indicator Redirection counter Redirecting reason No reply [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) ← 180 Ringing (Call-ID B-A, ACM) CFNR is performed ← INVITE(Call-ID B-C, IAM) Apply post test routine Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is the Redirection number present? Check: Is the Redirection information present and the Redirecting reason is set to 'No reply'? <th></th> <th colspan="4"></th>						
Redirection counter Redirecting reason No reply [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) + 180 Ringing (Call-ID B-A, ACM) CFNR is performed + INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is the Redirection number present and the Redirecting reason is set to 'No reply'?						
Redirecting reason No reply [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) + 180 Ringing (Call-ID B-A, ACM) CFNR is performed + INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is the Redirection number present? Check: Is Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to 'No reply'?						
No reply [any boundary name] Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) + 180 Ringing (Call-ID B-A, ACM) CFNR is performed + INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is Redirection number present? Check: Is Redirection number present? Check: Is Redirection number present?						
[any boundary name] Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) SIP (Network B) [any boundary name] INVITE(Call-ID A-B) → [any boundary name] [any boundary name] → [any boundary name] [any boundary name] [any boundary name] [any boundary name] [any boundary name] [any boundary name] Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirect						
Message flow SIP (Network A) Interconnection Interface INVITE(Call-ID A-B) SIP (Network B) ← 180 Ringing (Call-ID B-A, ACM) → CFNR is performed ← INVITE(Call-ID B-C, IAM) Apply post test routine Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the 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'?		No reply				
SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → (< 180 Ringing (Call-ID B-A, ACM) CFNR is performed (< INVITE(Call-ID B-C, IAM) Apply post test routine Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to 'No reply'?	Message flow	[any boundary name]				
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 an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to 'No reply'?		INVITE(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 an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the 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'?		CFNR is performed				
Comments Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to 'No reply'?						
 performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is the Redirection information present and the Redirecting reason is set to 'No reply'? 	Comments					
 Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to 'No reply'? 						
 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'? 						
 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'? 						
 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'? 						
 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'? 						
 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'? 						
Check: Is the Redirection number present?Check: Is Redirection information present and the Redirecting reason is set to 'No reply'?						
Check: Is Redirection information present and the Redirecting reason is set to 'No reply'?						
		Check: Is Redirection information present and the Redirecting reason is set to				
repeat this test in reverse direction.		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					
Test purpose	SIP-I support. CFNR performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.				
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded on no reply to user C,				
	user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.				
Configuration	Subscription options:				
5	 Served user releases his/her number to diverted-to user = Do not release 				
	diverting numberinformation				
SIP Parameter	INVITE				
	Content-Type: multipart/mixed;boundary=[any boundary name]				
	[any boundary name]				
	Content-Type: application/isup;version=itu-t92				
	Content-Disposition: signal;handling=required				
	IAM				
	Redirecting number				
	Address presentation restricted indicator				
	presentation restricted				
	Address signal (Diverting user)				
	Original called number				
	Address presentation restricted indicator				
	presentation restricted				
	Address signal				
	Redirection information				
	Original Redirection Reason				
	unknown				
	Redirecting indicator				
	Redirection counter				
	Redirecting reason				
	No reply				
Message flow	[any boundary name]				
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →				
	← 180 Ringing (Call-ID B-A, ACM)				
	CFNR is performed ← INVITE(Call-ID B-C, IAM)				
	Apply post test routine				
Comments	Originating user in Network A establishes a call to user in Network B. Network B				
	performs the diversion to a user in Network A.				
	Check: Is an INVITE request received at the interconnection interface?				
	Check: Is an IAM encapsulated in the INVITE?				
	Check: Is the Redirecting number present and the Address presentation				
	restricted indicator is set to 'presentation restricted'?				
	Check: Is the Original called number present and the Address presentation				
	restricted indicator is set to 'presentation restricted'?				
	Check: Is the Redirection number present?				
	Check: Is Redirection information present and the Redirecting reason is set to				
	'No reply'? Repeat this test in reverse direction.				
	וופריבו אווש נבשו ווו ובעבושב טוובנגוטוו.				

7.1.5.6.4 Communication Forwarding Not Logged in (CFNL)

		0.04				
Test case number	SS_cfnl_					
Test case group	SIP-SIP/	Service/CFNL				
Reference	4.5.2.6/ [9]					
SELECTION EXPRESSION	SE 28					
Test purpose	Communication forwarding not logged in, basic rules.					
	The user A and user C are in Network A. The user B is in network B and is provided with CFNL. Ensure that when user A calls user B, the call is forwarded not logged in to user C. In the active call state, ensure the property of speech.					
Configuration						
SIP Parameter						
Message flow SIP (Network A)	← ← ←	Interconnection Interface INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) ACK(Call-ID B-C) 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) Communication Apply post test routine	 → → → → 	SIP (Network B)		
Comments	Check: Check: Check: Repeat th	eck: The CDIV not logged in is successful. eck: In the active call state, ensure the property of speech.				

Test case number	SS_cfnl_0	SS_cfnl_002					
Test case group		Service/CFNL					
Reference	4.5.2.6/ [9)					
SELECTION EXPRESSION	SE 28 AN	D SE 30					
Test purpose	Commun	Communication forwarding not logged in, no notification.					
	provided v that his co Ensure th	The user A and user C are in Network A. The user B is in network B and is provided with CFNL, subscription option: "Originating user receives notification that his communication has been diverted" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, originating user is not notified.					
Configuration	Subscrip						
		munication has been					
SIP Parameter							
Message flow SIP (Network A)	(Interconnection Interface INVITE(Call-ID A-B) CFNL is performed INVITE(Call-ID B-C)	→	SIP (Network B)			
	(180 Ringing(Call-ID C-B) 180 Ringing(Call-ID B-A) Apply post test routine	→				
Comments	Check: No notification regarding call forwarding in network B is received at interconnection interface.						
	Repeat th	is test in reverse direction.					

Test case number	SS_cfnl_003	
Test case group	SIP-SIP/Service/CFNL	
Reference	4.5.2.6/[9]	
SELECTION EXPRESSION	SE 28 AND SE 30	
Test purpose	Communication forwarding not logged in, originating user is notified. URI	
rest purpose	of the diverted-to user not received.	
	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 = <mark>Yes</mark>	
	Served user allows the presentation of forwarded to URI to originating user	
	in diversion notification = No	
	 Served user allows the presentation of his/her URI to originating user in 	
	diversion notification = No	
SIP Parameter	181 Being Forwarded	
	P-Asserted-Identity: <userb@networkb></userb@networkb>	
	Privacy: id History-Info:	
	<sip:userb@networkb?<mark>Privacy=history>;index=1,</sip:userb@networkb?<mark>	
	<sip: userc@networka;cause="404?Privacy=history">;index=1.1</sip:>	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
	INVITE(Call-ID A-B) →	
	CFNL is performed	
	← INVITE(Call-ID B-C)	
	 181 Being Forwarded (Call-ID B-A) 	
	180 Ringing(Call-ID C-B) →	
	← 180 Ringing(Call-ID B-A)	
	Apply post test routine	
Comments	Check: A 181 Being Forwarded and a History-Info header is received at the	
	interconnection interface in both entries in the History-Info header a	
	Privacy header is escaped value 'history'.	
	Check: Is the cause parameter in the last entry is set to '404'?	
	Check: Is the "user=phone" parameter present in all History-Info header	
	URIS?	
	Check: Is the P-Asserted-Identity header present in the 181 identifying the served user?	
	NOTE: The history entries can be accumulated in "one" History-Info header or	
	each history entry is present in one single History-Info header.	
	Repeat this test in reverse direction.	

Test case number	SS_cfnl_004	
Test case group	SIP-SIP/Service/CFNL	
Reference	4.5.2.6/[9]	
SELECTION EXPRESSION	SE 28 AND SE 30	
Test purpose	Communication forwarding not logged in, originating user is notified. URI	
	from the diverted-to user received.	
	The user A and user C are in network A. The user B is in network B and is	
	provided with CFNL "Originating user receives notification that his	
	communication has been diverted" = Yes and "Served user allows the	
	presentation of forwarded to URI to originating user in diversion notification"	
	= Yes.	
	Ensure that when user A calls user B, the call is forwarded not logged in to user	
	C, user A is notified of call diversion and informed of the diverted-to number.	
Configuration	Subscription options:	
	 Originating user receives notification that his communication has been 	
	diverted = Yes	
	 Served user allows the presentation of forwarded to URI to originating user 	
	in diversion notification = Yes	
SIP Parameter	181 Being Forwarded	
	P-Asserted-Identity: <userb@networkb></userb@networkb>	
	History-Info:	
	<sip:userb@networkb>;index=1, <sip: userc@networka;cause="404">;index=1.1</sip:></sip:userb@networkb>	
Message flow	<sip. @networka,cause="404" usero="">,index=1.1</sip.>	
SIP (Network A)	Interconnection Interface SIP (Network B)	
	INVITE(Call-ID A-B) →	
	CFNL is performed	
	← INVITE(Call-ID B-C)	
	← 181 Being Forwarded(Call-ID B-A)	
	180 Ringing(Call-ID C-B) →	
	← 180 Ringing(Call-ID B-A)	
	Apply post test routine	
Comments	Check: A 181 Being Forwarded is received at interconnection interface.	
	Check: A History-Info header is contained in the 181 with the URI of the	
	served user and the URI of the diverted-to user.	
	Check: Is the cause parameter in the last entry is set to '404'?	
	Check: Is the "user=phone" parameter present in all History-Info header URIs?	
	Check: Is the P-Asserted-Identity header present in the 181 identifying the	
	served user?	
	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_	
Test case group	SIP-SIP/	Service/CFNL
Reference	4.5.2.6/ [91
SELECTION EXPRESSION		ND SE 30
Test purpose	Communication forwarding not logged in, diverted-to user does not	
		he 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
	•	to user" = No.
	Ensure th	nat when user A calls user B, the call is forwarded not logged in to user
		is not informed of the forwarding number.
Configuration		otion options:
		ed user allows the presentation of his/her URI to diverted-to user = No
SIP Parameter	INVITE	
	History-Ir	nfo:
		<pre>userB@networkB?Privacy=history>;index=1,</pre>
		userC@network1;cause=404>;index=1.1
Message flow		
SIP (Network A)		Interconnection Interface SIP (Network B)
		INVITE(Call-ID A-B) →
		CFNL is performed
	←	INVITE(Call-ID B-C)
		Apply post test routine
Comments	Check:	A History-Info header is received in the INVITE contains the URI of
		user B (served user) at the interconnection interface and a Privacy
		header is escaped set to 'history'.
	Check:	Is the cause parameter in the last entry is set to '404'?
	Check:	Is the "user=phone" parameter present in all History-Info header
		URIs?
	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 Request line may contain a 'cause' parameter indicating the
		redirecting reason.
	Repeat the	nis test in reverse direction.

Test case number	SS cfnl	006	
Test case group		Service/CFNL	
Reference	4.5.2.6/ [
SELECTION EXPRESSION			
Test purpose	SE 28 AND SE 30		
rest purpose	Communication forwarding not logged in, diverted-to user receives the URI of the served user.		
	of the se	ived user.	
	The user	A and user C are in network A. The user B is in network B and is	
		with CFNL "Served user allows the presentation of his/her URI to	
		to user" = Yes.	
		hat when user A calls user B, the call is forwarded not logged in to user	
		is informed of the forwarding number.	
Configuration		otion options:	
e egaranen		ed user allows the presentation of his/her URI to diverted-to user = Yes	
SIP Parameter	INVITE		
	History-Ir	nfo:	
		userB@networkB>;index=1,	
		userC@networkA;cause=404>;index=1.1	
Message flow	•		
SIP (Network A)		Interconnection Interface SIP (Network B)	
		INVITE(Call-ID A-B) →	
		CFNL is performed	
	÷	INVITE(Call-ID B-C)	
		Apply post test routine	
Comments	Check:	A History-Info header is received in the INVITE contains the URI of	
		user B (served user) at the interconnection interface.	
	Check:	Is the cause parameter in the last entry is set to '404'?	
	Check:	Is the "user=phone" parameter present in all History-Info header	
		URIs?	
	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 Request line may contain a 'cause' parameter indicating the	
	L .	redirecting reason.	
	Repeat th	nis test in reverse direction.	

Test case number	SS_cfnl_	007
Test case group		Service/CFNL
Reference	4.5.2.6/ [9]	
SELECTION EXPRESSION	SE 28 AN	
Test purpose		ication forwarding not logged in, full notification.
	The user provided communi of forward number is Ensure th C, user A	A and user C are in network A. The user B is in network B and is with CFNL "Originating user receives notification that his cation has been diverted" = Yes, "Served user allows the presentation ded to URI to originating user in diversion notification" = Yes, "diverting s released to the diverted-to user" = Yes. hat when user A calls user B, the call is forwarded not logged in to user is notified of call diversion and informed of the diverted-to number and
		informed of the forwarding number.
Configuration		tion options:
	 diver Servin diver 	nating user receives notification that his communication has been ted = Yes ed user allows the presentation of forwarded to URI to originating user version notification = Yes ting number is released to the diverted-to user = Yes
SIP Parameter	INVITE:	
	History-Ir <mark><sip:u< mark=""> <sip:< th=""><th>i<mark>serB@networkB&Reason=SIP;cause=404>;index=1</mark>, userC@networkA;cause=404>;index=1.1</th></sip:<></sip:u<></mark>	i <mark>serB@networkB&Reason=SIP;cause=404>;index=1</mark> , userC@networkA;cause=404>;index=1.1
	P-Asserte History-Ir <sip:u <sip:u 200 OK II History-Ir <sip:u< th=""><th>serB@network>;index=1, userC@networkA;cause=404>;index=1.1 NVITE</th></sip:u<></sip:u </sip:u 	serB@network>;index=1, userC@networkA;cause=404>;index=1.1 NVITE
Message flow SIP (Network A)		Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFNL is performed
	←	INVITE(Call-ID B-C)
		81 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)
	4	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
	Check:	interface sent to user C containing the URI identifying the served user. 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: Check:	Is the cause parameter in the last entry is set to '404'? Is the "user=phone" parameter present in all History-Info header URIs?
	Check: NOTE:	Is the P-Asserted-Identity header present in the 181 identifying the served user? The history entries can be accumulated in "one" History-Info header
	NOTE:	or each history entry is present in one single History-Info header. The Request line may contain a 'cause' parameter indicating the
	Repeat th	redirecting reason. is test in reverse direction.

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

Test case number	SS_cfnl_009		
Test case group	4.5.2.6/ [9]		
Reference	4.5.2.6/ [9]		
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)		
	 		
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here. Repeat this test in reverse direction.		

Test case number	SS cfnl 010	
Test case group	SIP-SIP/Service/CFNL	
Reference	4.5.2.6/ [9]	
SELECTION EXPRESSION	SE 28 AND SE 30 AND [Network A] SE 9	
Test purpose	Communication forwarding not logged in, interaction with a non trusted	
rest purpose	network.	
	The user A and user C are in network A. Network A is non trusted. 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" = No, "diverting number is released to the diverted-to user" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.	
Configuration	Subscription options:	
	 Originating user receives notification that his communication has been diverted = Yes Served user allows the presentation of forwarded to URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to originating user in diversion notification = No Served user allows the presentation of his/her URI to the diverted-to user = No 	
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFNL is performed ✓ INVITE(Call-ID B-C)	
	← 181 Being Forwarded(Call-ID B-A)	
	Apply post test routine	
Comments	Check: No History-Info header is received in the INVITE at the interconnection interface.	
	Check: No History-Info header is received in the 181 Being Forwarded at the interconnection interface (if sent).	
	Repeat this test in reverse direction.	

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

Test case number	SS_cfnl_012	
Test case group	SIP-SIP/Service/CFNL	
Reference	6.5/ [24]	
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 55C	
Test purpose	SIP-I support. Mobile subscriber not reachable 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 Mobile subscriber not reachable, 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 nama]	
	[any boundary name] Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	ACM Backward call indicator	
	Called party's status indicator	
	no indication	
	Redirection number	
	Address signal (Diverted-to user)	
	Call diversion information	
	Notification subscription options	
	presentation allowed without redirection number	
	Redirecting reason	
	Mobile subscriber not reachable Generic notification	
	call is diverting	
	[any boundary name]	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →	
	CFNL is performed	
	INVITE(Call-ID B-C, IAM)	
•	Harris 183 Session Progress (Call-ID B-A, ACM)	
Comments	Apply post test routine Originating user in Network A establishes a call to user in Network B. Network B	
Comments	performs the diversion to a user in Network A.	
	Check: 183 Session Progress is received at the interconnection interface.	
	Check: Is an ACM encapsulated in the 183?	
	Check: Is the Called party's status indicator set to 'no indication'?	
	Check: Is the Redirection number present?	
	Check: Is Notification subscription options indicator is set to 'presentation allowed without redirection number'?	
	Check: Is the Redirecting reason set to '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 [Network B] SE 47 AND SE 55C
Test purpose	SIP-I support. Mobile subscriber not reachable 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 Mobile subscriber not reachable, 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
Sir Falanietei	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 SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →
	 ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) Apply post test routine
Comments	Originating user in Network A establishes a call to user in Network B. Network B
	performs the diversion to a user in Network A.
	Check: 183 Session Progress is received at the interconnection interface Check: Is an ACM encapsulated in the 183?
	Check: Is the Called party's status indicator set to 'no indication'?
	Check: Is the Redirection number present? Check: Is Notification subscription options indicator is set to 'presentation
	allowed with redirection number'? Check: Is the Redirecting reason set to 'Mobile subscriber not reachable'? Repeat this test in reverse direction.

Test case number	SS_cfnl_014		
Test case group	SIP-SIP/Service/CFNL		
Reference	6.7/ [24]		
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 53 AND SE 55C		
Test purpose	SIP-I support. Mobile subscriber not reachable 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 Mobile subscriber not reachable, 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		
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		
SIF Farameter	Content-Type: multipart/mixed;boundary=[any boundary name]		
	Contone Typo: manparemixed, boundary=[any boundary hamo]		
	[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) →		
	CFNL is performed		
	← INVITE(Call-ID B-C, IAM)		
	180 Ringing (Call-ID C-B) →		
	← 180 Ringing (Call-ID B-A, ACM)		
	200 OK INVITE (Call-ID C-B) →		
	← ACK (Call-ID B-C) ← 200 OK INVITE (Call-ID B-A ANM)		
	← 200 OK INVITE (Call-ID B-A, ANM) ACK (Call-ID A-B) →		
	Apply post test routine		
Comments	Originating user in Network A establishes a call to user in Network B. Network B		
	performs the diversion to a user in Network A.		
	Check: Is a 200 OK INVITE received at the interconnection interface		
	Check: Is an ANM encapsulated in the 200 OK?		
	Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation		
	restricted"?		
	Repeat this test in reverse direction.		

Test case number	SS_cfnl_015				
Test case group	SIP-SIP/Service/CFNL				
Reference	6.7/ [24]				
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 53 AND SE 55C				
Test purpose	SIP-I support. Mobile subscriber not reachable performed in Network B, No				
rest purpose	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 Mobile subscriber not reachable, 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:				
0	 Connected user subscribed to COLR = no 				
SIP Parameter	200 OK				
	Content-Type: multipart/mixed;boundary=[any boundary name]				
	[any boundary name]				
	Content-Type: application/isup;version=itu-t92				
	Content-Disposition: signal;handling=required				
	ANM				
	Redirection number restriction				
	Presentation allowed				
	or				
	Redirection number restriction not present				
	[any boundary name]				
Message flow					
SIP (Network A)	Interconnection Interface SIP (Network B)				
	INVITE(Call-ID A-B) →				
	CFNL is performed				
	← INVITE(Call-ID B-C, IAM)				
	180 Ringing (Call-ID C-B) →				
	← 180 Ringing (Call-ID B-A, ACM)				
	200 OK INVITE (Call-ID C-B) →				
	← ACK (Call-ID B-C)				
	← 200 OK INVITE (Call-ID B-A, ANM)				
	ACK (Call-ID A-B) →				
Commonto	Apply post test routine				
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A.				
	Check: Is a 200 OK INVITE received at the interconnection interface?				
	Check: Is a 200 OK INVITE received at the interconnection interface? Check: Is an ANM encapsulated in the 200 OK?				
	Check: Is the ISUP/BICC Redirection number restriction present set to				
	'Presentation allowed' or is the parameter absent? Repeat this test in reverse direction.				

Test case number	SS_cfnl_016				
Test case group	SIP-SIP/Service/CFNL				
Reference	7.1/ [24]				
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 55C				
Test purpose	 SIP-I support. Mobile subscriber not reachable 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 Mobile subscriber not reachable, Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B. 				
Configuration	Subscription options:				
	 Served user releases his/her number to diverted-to user = Release diverting 				
	number information				
SIP Parameter	INVITE				
	Content-Type: multipart/mixed;boundary=[any boundary name]				
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (<i>Diverting user</i>) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason				
	unknown				
	Redirecting indicator				
	Redirection counter				
	Redirecting reason Mobile subscriber not reachable				
	[any boundary name]				
Message flow 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 an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present and the Address presentation restricted indicator is set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is Redirection information present and the Redirecting reason is set to '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 [Network B] SE 47 AND SE 55C					
Test purpose	SIP-I support. Mobile subscriber not reachable 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 Mobile subscriber not reachable, 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.					
	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:					
g	 Served user releases his/her number to diverted-to user = Do not release diverting numberinformation 					
SIP Parameter						
	Content-Type: multipart/mixed;boundary=[any boundary name]					
	[any boundary name] Content-Type: application/isup;version=itu-t92					
	Content-Disposition: signal;handling=required					
	IAM					
	Redirecting number					
	Address presentation restricted indicator					
	presentation restricted Address signal (<i>Diverting user</i>)					
	Original called number					
	Address presentation restricted indicator					
	presentation restricted					
	Address signal					
	Redirection information					
	Original Redirection Reason					
	unknown					
	Redirecting indicator					
	Redirection counter					
	Redirecting reason Mobile subscriber not reachable					
Message flow	[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) Apply post test routine					
Comments	Originating user in Network A establishes a call to user in Network B. Network B					
	performs the diversion to a user in Network A.					
	Check: Is an INVITE request received at the interconnection interface?					
	Check: Is an IAM encapsulated in the INVITE?					
	Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation 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.					

Test case number	SS_cd_0	001				
Test case group	SIP-SIP/Service/CD					
Reference	4.5.2.6/ [9]				
SELECTION EXPRESSION	SE 29					
Test purpose	Commu	nication deflection during alertin	ng, basic r	ules.		
		A and user C are in Network A. Th	he user B i	s in network B and is		
	P	with CDa.				
		hat when user A calls user B, the c				
	user C. I	n the active call state, ensure the p	property of	speech.		
Configuration						
SIP Parameter						
Message flow						
SIP (Network A)			•	SIP (Network B)		
		INVITE(Call-ID A-B)	→			
	CDa is performed ← 180 Ringing(Call-ID B-A) ← INVITE(Call-ID B-C)					
	✓ INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B)					
	← 180 Ringing(Call-ID B-A)					
	•	200 OK INVITE(Call-ID C-B) →				
	←	ACK(Call-ID B-C)				
	÷	200 OK INVITE(Call-ID B-A)				
		ACK(Call-ID A-B)	→			
		Communication				
		Apply post test routine				
Comments	Check: CDa is successful.					
	Check: In the active call state, ensure the property of speech.					
	Check: Is the P-Asserted-Identity present in the INVITE sent from Network B					
	to Network A set to the identity of the originating user?					
	Repeat t	his test in reverse direction.				

7.1.5.6.5 Communication Deflection

Test case number	SS_cd_0	02			
Test case group	SIP-SIP/	SIP-SIP/Service/CD			
Reference	4.5.2.6/ [9]			
SELECTION EXPRESSION	SE 29				
Test purpose	Commu	nication deflection immediate, b	asic rules.		
	-				
		A and user C are located in Netw	•••••		
		B and is provided with CDi. Ensure			
		mmediately the communication to ne call is forwarded to user C. In the			
		of speech.		in state, ensure the	
Configuration	property				
SIP Parameter					
Message flow					
SIP (Network A)		Interconnection Interface		SIP (Network B)	
		INVITE(Call-ID A-B)	→	on (nonen 2)	
		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			
0		Apply post test routine			
Comments	•••	Check: CDi is successful.			
	Check:	Check: In the active call state, ensure the property of speech. Check: Is the P-Asserted-Identity present in the INVITE sent from Network B			
	Check:	to Network A set to the identity of			
	Reneat th	his test in reverse direction.			
	inepeat ti				

Test case number	SS cd 003			
Test case group	SIP-SIP/Service/CD			
Reference	4.5.2.6/ [9]			
SELECTION EXPRESSION	SE 29 AND SE 30			
Test purpose	Communication Deflection immediate response, no notification.			
	The user A and user C are located in Network A. The user B is located in network B and is provided with CDi, subscription option: Originating use receives notification that his communication has been diverted = No. Ensure that when user A calls user B which deflects immediately the communication towards user C (i.e. before alerting starts), the call is for to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded mes	r warded		
Configuration	Subscription options:	<u></u>		
	 Originating user receives notification that his communication has been diverted = No 			
SIP Parameter				
Message flow SIP (Network A)	Interconnection Interface SIP (Network INVITE(Call-ID A-B) → CDi is performed ► INVITE(Call-ID B-C)	а В)		
	 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 receiv	ed at the		
	interconnection interface.			
	Check: Is the cause parameter in the last entry is set to '480'?			
	Repeat this test in reverse direction.			

Test sees number		04		
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 located in network A. The user B is located in network B and is provided with CDi "Originating user receives notification that his communication has been diverted" = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B which deflects immediately the communication towards user C (i.e. before alerting starts), the call is forwarded to user C.			
	Ensure th	nat User A receives a 181 Call Is Being Forwarded message, user A is f call diversion and not informed of the diverted-to number and served		
Configuration	Subscrip	otion options:		
	• Orig	inating user receives notification that his communication has been		
		rted = <mark>Yes</mark>		
	• Orig	inating user receives notification that his communication has been		
		rted = No		
		red user allows the presentation of his/her URI to originating user in rsion notification = No		
SIP Parameter	181 Bein	g Forwarded		
	P-Assert	ed-Identity: <userb@networkb></userb@networkb>		
	Privacy:	d		
	History-Ir	nfo:		
		userB@networkB? <mark>Privacy=history</mark> >;index=1,		
	<sip: userc@networka;cause="480?Privacy=history">;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)		
		81 Being Forwarded (Call-ID B-A)		
		Apply post test routine		
Comments	Check:	A 181 Being Forwarded and a History-Info header is received at the		
Comments	oncon.	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'?		
	Check:	Is the "user=phone" parameter present in all History-Info header		
	URIS?			
	Check:	Is the P-Asserted-Identity header present in the 181 identifying the served user?		
	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 t	his test in reverse direction.		

Test case number	SS_cd_005			
Test case group	SIP-SIP/Service/CD			
Reference	4.5.2.6/ [9]			
SELECTION EXPRESSION	SE 29 AND SE 30			
Test purpose	Communication Deflection immediate response, originating user is			
	notified. URI from the diverted-to user received.			
	The user A and user C are in network A. The user B is in network B and is			
	provided with CDi "Originating user receives notification that his communication			
	has been diverted" = Yes and "Served user allows the presentation of forwarded			
	to URI to originating user in diversion notification" = Yes.			
	Ensure that when user A calls user B which deflects immediately the			
	communication towards user C (i.e. before alerting starts), the call is forwarded			
	to user C.			
	Ensure that User A receives a 181 Call Is Being Forwarded message, user A is			
Configuration	notified of call diversion and informed of the diverted-to number. Subscription options:			
Configuration	 Originating user receives notification that his communication has been 			
	 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			
	P-Asserted-Identity: <userb@networkb></userb@networkb>			
	History-Info:			
	<sip:userb@networkb>;index=1,</sip:userb@networkb>			
	<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)			
	 181 Being Forwarded (Call-ID B-A) 			
Comments	Apply post test routine Check: A 181 Being Forwarded is received at the interconnection interface.			
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'?			
	Check: Is the "user=phone" parameter present in all History-Info header			
	URIS?			
	Check: Is the P-Asserted-Identity header present in the 181 identifying the			
	served user?			
	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 CDi "Served user allows the presentation of his/her URI to the diverted-to user" = No.			
	Ensure that when user A calls user B which deflects immediately the communication towards user C (i.e. before alerting starts), the call is forwarded to user C, user C is not informed of the forwarding number.			
Configuration	 Subscription options: Served user allows the presentation of his/her URI to diverted-to user = No 			
SIP Parameter	INVITE History-Info: <sip:userb@networkb?privacy=history&reason=sip%3bcause%3d302> ;index=1, <sip: userc@networka;cause="480">;index=1.1</sip:></sip:userb@networkb?privacy=history&reason=sip%3bcause%3d302>			
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'.			
	 heck: Is the cause parameter in the last entry is set to '480'? heck: Is the "user=phone" parameter present in all History-Info header URIs? 			
	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 Request line may contain a 'cause' parameter indicating the			
	redirecting reason. Repeat this test in reverse direction.			

Test case number	SS_cd_007			
Test case group	SIP-SIP/Service/CD			
Reference	4.5.2.6/ [9]			
SELECTION EXPRESSION	SE 29 AND SE 30			
Test purpose	Communication Deflection immediate response, diverted-to user receives the URI of the served user.			
	The user A and user C are in network A. The user B is in network B and is provided with CDi "Served user allows the presentation of his/her URI to diverted-to user" = Yes. Ensure that when user A calls user B which deflects immediately the communication towards user C (i.e. before alerting starts), the call is forwarded to user C, user C is informed of the forwarding number.			
Configuration	Subscription options:			
g	 Served user allows the presentation of his/her URI to diverted-to user = Yes 	s		
SIP Parameter	INVITE History-Info: <sip:userb@networkb?reason=sip%3bcause%3d302>;index=1, <sip: userc@networka;cause="480">;index=1.1</sip:></sip:userb@networkb?reason=sip%3bcause%3d302>			
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CDi is performed INVITE(Call-ID B-C) Apply post test routine			
Comments	Check: A History-Info header is received in the INVITE contains the URI of user B (served user) at the interconnection interface.			
	Check: Is the cause parameter in the last entry is set to '480'?			
	Check: Is the "user=phone" parameter present in all History-Info header URIs?			
	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 Request line may contain a 'cause' parameter indicating the redirecting reason.			
	Repeat this test in reverse direction.			

Test case number	SS_cd_0	08		
Test case group		Service/CD		
Reference	4.5.2.6/ [9			
SELECTION EXPRESSION	SE 29	-1		
Test purpose	Commun	ication Deflection immediate re	esponse, ur	nsuccessful UDUB.
	provided			
		at when user A calls user B, the	call is deflec	ted 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 re	eceiving a 48	36 Busy Here.
	Repeat th	is test in reverse direction.	Ũ	-

Test case number	SS_cd_0	009		
Test case group	SIP-SIP/	Service/CD		
Reference	4.5.2.6/	9]		
SELECTION EXPRESSION	SE 29	-		
Test purpose	Commu	nication Deflection immediate re	esponse, ι	Insuccessful NDUB.
	Ensure t	A and user C are in network A. The two in two in the two in two in two in the two in t	call is defle	
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	ceiving a 4	186 Busy Here.
	Repeat t	his test in reverse direction.	5	

Test case number	SS_cd_010			
Test case group	SIP-SIP/Service/CD			
Reference	4.5.2.6/ [9]			
SELECTION EXPRESSION	SE 29 AND SE 30 AND [Network A] SE 9			
Test purpose	Communication Deflection immediate response, interaction with a non trusted network.			
	The user A and user C are in network A. Network A is non trusted. The user B is in network B and is provided with CDi "Originating user receives notification that his communication has been diverted" = Yes "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No, "diverting number is released to the diverted-to user" = No. Ensure that when user A calls user B, the call is deflected immediate response to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.			
Configuration	Ŭ			
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 [Network B] SE 47 AND SE 55D		
Test purpose	SIP-I support. CD performed in Network B, Notification subscription options is set to presentation not allowed.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is deflected to user C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM or CPG contained in the Redirection number and Call diversion information if SIP-I -		
	ISUP/BICC interworking is applicable in Network B.		
Configuration	Subscription options:		
	 Calling user receives notification that his call has been diverted (forwarded 		
	or deflected) = <mark>no</mark>		
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 not allowed Redirecting reason Deflection immediate or Deflection during alerting Generic notification call is diverting		
	[any boundary name]		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE(Call-ID A-B) → 180 Ringing (Call-ID B-A, ACM) in case CDa CD is performed INVITE(Call-ID B-C, IAM) 183/180 (Call-ID B-A, ACM/CPG) Apply post test routine		
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is a 183 Session Progress received at the interconnection interface? Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator set to 'presentation not allowed'? Check: Is the Redirecting reason set to 'Deflection immediate' or 'Deflection during alerting'? Repeat this test in reverse direction.		

Test case number	SS_cd_012		
Test case group	SIP-SIP/Service/CD		
Reference	6.5/ [24]		
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 55D		
Test purpose	SIP-I support. CD performed in Network B, Notification subscription options is set to presentation allowed without redirection number.		
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number. Ensure that when user A calls user B, the call is deflected to user C, user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM or CPG		
	contained in the Redirection number and Call diversion information if SIP-I - ISUP/BICC interworking is applicable in Network B.		
Configuration	Subscription options:		
	 Calling user receives notification that his call has been diverted (forwarded 		
	or deflected) = yes, without diverted-to user number		
SIP Parameter	183 /180 Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	ACM/CPG Redirection number Address signal (<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 boundary name]		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 180 Ringing (Call-ID B-A, ACM) in case CDa → CD is performed INVITE(Call-ID B-C, IAM) € 183 / 180 (Call-ID B-A, ACM/CPG) Apply post test routine 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: 18 an ACM encapsulated in the 183? Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator is set to 'presentation allowed without redirection number'? Check: Is the Redirecting reason set to 'Deflection immediate' or 'Deflection during alerting'? Repeat this test in reverse direction.		

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

Test case number	SS_cd_014			
Test case group	SIP-SIP/Service/CD			
Reference	6.7/[24]			
SELECTION EXPRESSION				
	[Network B] SE 17 AND [Network B] SE 47 AND SE 53 AND SE 55D			
Test purpose	SIP-I support. CD performed in Network B, Restriction of the Redirectio number.			
	number.			
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part			
	of Network B and is provided with CDi or CDa, Diverted-to user is subscribed to			
	the COLR service in Permanent mode.			
	Ensure that when user A calls user B, the call is deflected to user C, a			
	Redirection number restriction parameter is present set to 'Presentation			
	restricted' in the encapsulated ANM contained in the 200 OK INVITE if			
	ISUP/BICC- SIP-I interworking is applicable in Network A.			
Configuration	Subscription options:			
••····g	 Connected user subscribed to COLR, Permanent = yes 			
SIP Parameter	200 OK			
	Content-Type: multipart/mixed;boundary=[any boundary name]			
	[any boundary name]			
	Content-Type: application/isup;version=itu-t92			
	Content-Disposition: signal;handling=required			
	ANM			
	Redirection number restriction			
	Presentation restricted			
	[any boundary name]			
Message flow				
SIP (Network A)	Interconnection Interface SIP (Network B)			
	INVITE(Call-ID A-B) →			
	 180 Ringing (Call-ID B-A, ACM) in case CDa 			
	CD is performed			
	← INVITE(Call-ID B-C, IAM)			
	180 Ringing (Call-ID C-B) →			
	← 180 Ringing (Call-ID B-A, ACM/CPG)			
	200 OK INVITE (Call-ID C-B) →			
	← ACK (Call-ID B-C) ← 200 OK INVITE (Call-ID B-A ANM)			
	E 200 OK INVITE (Call-ID B-A, ANM) ACK (Call-ID A-B) →			
	Apply post test routine			
Comments	Originating user in Network A establishes a call to user in Network B. Network B			
Comments	performs the diversion to a user in Network A.			
	Check: Is a 200 OK INVITE received at the interconnection interface?			
	Check: Is a 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 B] SE 17 AND [Network B] SE 47 AND SE 53 AND SE 55D				
Test purpose	SIP-I support. CD performed in Network B, No restriction of the Redirection				
	number.				
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part				
	of Network B and is provided with CDi or CDa, Diverted-to user is not subscribed				
	to the COLR service.				
	Ensure that when user A calls user B, the call is deflected to user C, if a				
	Redirection number restriction parameter is present it is set to 'Presentation				
	allowed' in the encapsulated ANM contained in the 200 OK INVITE if				
	ISUP/BICC- SIP-I interworking is applicable in Network A.				
Configuration	Subscription options:				
	Connected user subscribed to COLR = no				
SIP Parameter	200 OK				
	Content-Type: multipart/mixed;boundary=[any boundary name]				
	[any boundary name]				
	Content-Type: application/isup;version=itu-t92				
	Content-Disposition: signal;handling=required				
	ANM				
	Redirection number restriction				
	Presentation allowed				
	or				
	Redirection number restriction not present				
	[any boundary name]				
Message flow					
SIP (Network A)	Interconnection Interface SIP (Network B)				
	INVITE(Call-ID A-B) →				
	← 180 Ringing (Call-ID B-A, ACM) in case CDa				
	CD is performed				
	← INVITE(Call-ID B-C, IAM)				
	180 Ringing (Call-ID C-B) →				
	← 180 Ringing (Call-ID B-A, ACM/CPG)				
	200 OK INVITE (Call-ID C-B) →				
	← ACK (Call-ID B-C)				
	← 200 OK INVITE (Call-ID B-A, ANM)				
	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 [Network B] SE 47 AND SE 55D				
Test purpose	SIP-I support. CD performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.				
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Served user releases his/her number to diverted-to user = Release diverting number information.				
	Ensure that when user A calls user B, the call is deflected to user C, user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if				
	ISUP/BICC - SIP-I interworking is applicable in Network B.				
Configuration	 Subscription options: Served user releases his/her number to diverted-to user = Release diverting 				
	 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				
	Deflection immediate or Deflection during alerting				
	[any boundary name]				
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) →				
	 ← 180 Ringing (Call-ID B-A, ACM) in case CDa CD is performed 				
	INVITE(Call-ID B-C, IAM) Apply post test routine				
Comments	Originating user in Network A establishes a call to user in Network B. Network B				
	performs the diversion to a user in Network A.				
	Check: Is an INVITE request received at the interconnection interface?				
	Check: Is an IAM encapsulated in the INVITE?				
	Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'?				
	Check: Is the Original called number present and the Address presentation				
	restricted indicator is set to 'presentation allowed'?				
	Check: Is the Redirection number present?				
	Check: Is Redirection information present and the Redirecting reason is set to				
	'Deflection immediate' or 'Deflection during alerting'?				
	Repeat this test in reverse direction.				

Test case number	SS_cd_017				
Test case group	SIP-SIP/Service/CD				
Reference	7.1/ [24]				
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 55D				
Test purpose	SIP-I support. CD performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.				
	The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Served user releases his/her				
	number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is deflected to user C, user C is				
	notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if ISUP/BICC - SIP-I interworking is applicable in Network B.				
Configuration	Subscription options:				
Comgulation	 Served user releases his/her number to diverted-to user = Do not release 				
	diverting numberinformation				
SIP Parameter	INVITE				
	Content-Type: multipart/mixed;boundary=[any boundary name]				
	[any boundary name]				
	Content-Type: application/isup;version=itu-t92				
	Content-Disposition: signal;handling=required				
	IAM				
	Redirecting number				
	Address presentation restricted indicator				
	presentation restricted				
	Address signal (<i>Diverting user</i>) Original called number				
	Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown				
	Redirecting indicator				
	Redirection counter				
	Redirecting reason				
	Deflection immediate or Deflection during alerting				
Magazza (law	[any boundary name]				
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)				
	INVITE(Call-ID A-B) → ◆ 180 Ringing (Call-ID B-A, ACM) in case CDa				
	CD is performed ← INVITE(Call-ID B-C, IAM)				
	Apply post test routine				
Comments	Originating user in Network A establishes a call to user in Network B. Network B				
	performs the diversion to a user in Network A.				
	Check: Is an INVITE request received at the interconnection interface?				
	Check: Is an IAM encapsulated in the INVITE?Check: Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation 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.				

7.1.5.6.6 Call establishment when multiple diversions occur

Test case number	SS multi	ipleCFU_001		
Test case group	SIP-SIP/Service/multibleCF			
Reference	4.5.2.6/ [9]			
SELECTION EXPRESSION	SE 17 AND SE 22			
Test purpose	Call establishment with multiple forwarding user is not informed. The user A and user C are in network A. The user B and user D are in network B. The user B and user C are provided with CFU "Served user allows the presentation of his/her URI to diverted-to user" = No.			
	C and us numbers	Ensure that when user A calls user B, the call is forwarded unconditional to user C and user C forwards to user D, and user D is not informed of the forwarding numbers.		
Configuration		tion options: ved user allows the presentation of his/her URI to diverted-to user" = <mark>No</mark>		
SIP Parameter	INVITE: History-Info: <sip; userb@networkb?privacy="history">;index=1, <sip:userc@networka;cause=302>;index=1.1,</sip:userc@networka;cause=302></sip;>			
	INVITE: History-Info: <sip; userb@networkb?privacy="history">;index=1, <sip:userc@networka;cause=302?privacy=history>;index=1.1, <sip:userd@networkb;cause=302>;index=1.1.1</sip:userd@networkb;cause=302></sip:userc@networka;cause=302?privacy=history></sip;>			
Message flow				
SIP (Network A)		Interconnection Interface SIP (Network B)		
	÷	INVITE(Call-ID A-B) → CFU is performed INVITE(Call-ID B-C) CFU is performed INVITE(Call-ID C-D) → Apply post test routine		
Comments	Check:	Is a History-Info header containing Index number 1 present in the INVITE from User C to user D?		
	Check:	Is a History-Info header containing Index number 1.1 present in the INVITE from User C to user D?		
	Check:	Is a History-Info header containing Index number 1.1.1 present in the INVITE from User C to user D?		
	Check:	Does the History-Info header index 1 received in the INVITE contain the URI of user B (first served user) at the interconnection interface and a Privacy header is escaped set to 'history'?		
	Check: Does the History-Info header index 1.1 received in the INVITE contain the URI of user C (second served user) at the interconnection interface and a Privacy header is escaped set to 'history'?			
	Check:	Does the History-Info header index 1.1.1 received in the INVITE contain the URI of user D (Terminating user) at the interconnection interface?		
	Check: Is the cause parameter in the last two entries of the History-Info Header set to '302'?			
	Check: Is the "user=phone" parameter present in all History-Info header URIs?			
	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 Request line may contain a 'cause' parameter indicating the redirecting reason.			
		his test in reverse direction.		

Test case number	SS_multi	pleCFU_002		
Test case group	SIP-SIP/Service/multibleCF			
Reference	4.5.2.6/ [9]			
SELECTION EXPRESSION	SE 17 AND SE 22			
Test purpose	Call establishment with multiple forwarding.			
	The user B. The us	The user A and user C are in network A. The user B and user D are in network B. The user B and user C 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 and user C forwards to user D, and user D will be informed of the forwarding numbers		
Configuration	Subscript	ion options:		
	 "Servention of the servention of the serventio oo the servention of the servention oo the servention oo the	ved user allows the presentation of his/her URI to diverted-to user" =		
SIP Parameter	INVITE:			
		ifo header:		
		iserB@networkB>;index=1, iserC@networkA;cause=302>;index=1.1,		
	INVITE:			
		ifo header:		
	<sip:u< th=""><th>IserB@networkB>;index=1,</th></sip:u<>	IserB@networkB>;index=1,		
		<pre>serC@networkA;cause=302>;index=1.1,</pre>		
Message flow	<u> </u>	<pre>iserD@networkB;cause=302>;index=1.1.1</pre>		
SIP (Network A)		Interconnection Interface SIP (Network B)		
	÷	INVITE(Call-ID A-B) → CFU is performed INVITE(Call-ID B-C) CFU is performed INVITE(Call-ID C-D) → Apply post test routine		
Comments	Check:	Is a History-Info header containing Index number 1 present in the INVITE from User C to user D?		
	Check:	Is a History-Info header containing Index number 1.1.1 present in the		
		INVITE from User C to user D?		
	Check: Does the History-Info header index 1 received in the INVITE contain the URI of user B (first served user) at the interconnection interface?			
	Check: Does the History-Info header index 1.1 received in the INVITE contain the URI of user C (second served user) at the interconnection interface?			
	Check:	Does the History-Info header index 1.1.1 received in the INVITEcontain the URI of user D (Terminating user) at the interconnection interface?		
	Check:	Is the cause parameter in the last two entries of the History-Info Header set to '302'?		
	Check:	Is the "user=phone" parameter present in all History-Info header URIs?		
	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 Request line may contain a 'cause' parameter indicating the redirecting reason.			
	Repeat th	is 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 'Trying'. 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 'Trying'. The user B2 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 'Trying'. The user B2 sends an INVITE request to the conference focus in network A. Is the request is confirmed, user B2 sends a NOTIFY indicating '200 OK'. User A terminates the original dialogue. 		
Configuration			
SIP Parameter	REFER(user B1) Refer-To: <uri conference="" focus;method="INVITE" of=""> NOTIFY(B1, 100) Content-Type: message/sipfrag SIP/2.0 100 INVITE: Request URI: uri of conference focus From: user B1 NOTIFY(B1, 200) Content-Type: message/sipfrag SIP/2.0 200 OK REFER(user B2) Refer-To: <uri conference="" focus;method="INVITE" of=""> NOTIFY(B2, 100) Content-Type: message/sipfrag SIP/2.0 100 INVITE: Request URI: uri of conference focus From: user B2 NOTIFY(B2, 200) Content-Type: message/sipfrag SIP/2.0 200 OK</uri></uri>		

Message flow			
SIP (Network A)		Interconnection Interface	SIP (Network B)
		to user B1 from Network A to N	
Establish a confirmed	d session	to user B2 from Network A to N	Network B and put it on hold
	User /	A establishes a 3PTY conversat	tion
		REFER(user B1)	→
	←	202 Accepted	
	←	NOTIFY(B1, 100)	
		200 OK NOTIFY	→
	÷	INVITE(focus, user B1)	
		200 INVITE	\rightarrow
	←	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	after the confirmed communication to
	user B1	and B2 was set on HOLD	
	Check:	The Refer-To header in the RE	FER method sent to user B1 and B2
		contains the URI of the conferen	nce focus and is the method parameter
		set to 'INVITE'.	
	Check:	The NOTIFY after the REFER r	equest contains the 'SIP/2.0 100'
		message body.	
	Check:	The INVITE request is sent by u	user B1 and user B2 to the conference
			from the Refer-To header of the
		received REFER request.	
	Check:		equest contains the 'SIP/2.0 200 OK'
		message body.	•
	Check:	The original session is terminate	ed by user A.
	Repeat t	his test in reverse direction.	

Test case number	SS_conf_002
Test case group	SIP-SIP/Service/CONF
Reference	4.5.2/ [10], 4.7.2.9.7/ [20]
SELECTION EXPRESSION	[Network A] SE 12 AND SE 31
Test purpose	3 Party establishment using reINVITE performed by the AS in network A.
	o raity establishment using rentitie performed by the Ao 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
	relNVITE request.
	Ensure that user A can invite user B2 to the conference by sending a
	relNVITE request.
Configuration	
SIP Parameter	INVITE <b1></b1>
	From: <usera></usera>
	To: <userb1></userb1>
	Call-ID: A-B1
	P-Asserted-Identity: <usera></usera>
	SDP: a=sendrecy
	INVITE <b2></b2>
	From: <usera></usera>
	Call-ID: A-B2
	To: <userb2></userb2>
	P-Asserted-Identity: <usera></usera>
	······································
	SDP: a=sendrecv
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
Establish a confirme	ed session to user B1 from Network A to Network B and put it on hold
Establish a confirme	ed session to user B2 from Network A to Network B and put it on hold
	User A establishes a 3PTY conversation
	INVITE(Call-ID A-B1) →
	← 200 INVITE
	ACK 🔸
	INVITE(Call-ID A-B2) →
	► 200 INVITE
	ACK →
	Apply post test routine
Comments	User A establishes a 3PTY conversation after the confirmed communication to
	user B1 and B2 was set on HOLD
	Check: An INVITE is sent to user B1 and user B2 indicating a new IP address
	in the 'c' line of the SDP.
	Check: The 'a' line indicates 'sendrecv'.
	Repeat this test in reverse direction.

Test case number	SS_conf_003
Test case group	SIP-SIP/Service/CONF
Reference	5.4/ [24]
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] 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 an ISUP/BICC
	CPG is encapsulated the Generic Notification is set to 'conference
	established';
	 an INFO request is sent to User B2 in Network B and an 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
	000
	CPG Generic Notification
	Conference established
	Conterence established
	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)	Interconnection Interface SIP (Network B)
	ssion from User A in Network A to user B1 in Network B and put it on hold
Establish a con	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 an ISUP/BICC CPG message encapsulated in the INFO request to
a	Check. Is all ISOF/DICC CFG message encapsulated in the INFO request to
	both remote users in Network B?
	both remote users in Network B?

Test case number	SS_conf_004	
Test case group	SIP-SIP/Service/CONF	
Reference		
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] SE 47 AND SE 56	
Test purpose	SIP-I/ISUP interworking. Served user disconnects one of the remote use	rs.
	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 U	ser
	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	on.
	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 an ISUP/BICC	
	CPG is encapsulated the Generic Notification is set to 'Conference	
	disconnected'.	
Configuration	ISUP/BICC interworking applies in Network A	
	User in Network A is subscribed to the 3PTY supplementary service	
SIP Parameter	INFO <b2></b2>	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	CPG	
	Generic Notification	
	Conference disconnected	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
Ectablich a contirmed acc		
	sion from User A in Network A to user B1 in Network B and put it on hold	
	rmed session from User A in Network A to user B2 in Network B	
	rmed session from User A in Network A to user B2 in Network B User A establishes a 3PTY conversation	
	rmed 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) →	
	rmed session from User A in Network A to user B2 in Network B User A establishes a 3PTY conversation	
	Image: Series of the serie	
	rmed 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) →	
	Immed 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	
Establish a confi	Immed 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	
	rmed 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 E 200 INFO Apply post test routine User A establishes a 3PTY conversation with user B1 and user B2 located in	
Establish a confi	rmed 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 Production (Call-ID A-B2, CPG) → 200 INFO Apply post test routine User A establishes a 3PTY conversation with user B1 and user B2 located in Network B.	
Establish a confi	Image: Session from User A in Network A to user B2 in Network B User A establishes a 3PTY conversation BYE(Call-ID A-B1, REL) ← 200 INFO ▲ 200 INFO ▲ 200 INFO ▲ 200 INFO ▲ By E(Call-ID A-B2, CPG) ★ 200 INFO ▲ Apply post test routine 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 communication with user B1 in Network B)	
Establish a confi	Image: 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 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 of hold).	
Establish a confi	Image: 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 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 chold). Check: Is a BYE request is sent to user B1 in Network B?	on
Establish a confi	Image: 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) Image: Conversation BYE(Call-ID A-B1, REL) 200 INFO Image: Conversation Image: Conversation BYE(Call-ID A-B2, CPG) Image: Conversation Image: Co	on
Establish a confi	rmed 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 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 of hold). Check: Is a BYE request is sent to user B1 in Network B? Check: Is an ISUP/BICC CPG message encapsulated in the INFO request user B2 in Network B?	on to
Establish a confi	 Impose the provided and the imposed and the provided and the imposed and the provided and the	on to
Establish a confi	rmed 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 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 of hold). Check: Is a BYE request is sent to user B1 in Network B? Check: Is an ISUP/BICC CPG message encapsulated in the INFO request user B2 in Network B?	on to

Test case number	SS_conf_005	
Test case group	SIP-SIP/Service/CONF	
Reference	5.4/[24]	
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] SE 47 AND SE 56	
Test purpose	SIP-I/ISUP interworking. Served user splits the 3 Party communication.	
rest purpose		
	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 an 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 an 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) Establish a confirmed session from User A in Network A to user B1 in Network B and put it on hold Establish a confirmed session from User A in Network A to user B2 in Network B User A establishes a 3PTY conversation INFO(Call-ID A-B1, CPG) →		
	← 200 INFO	
	INFO(Call-ID A-B2, CPG) → € 200 INFO	
Comments	Apply post test routine 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 an 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 [Network A] 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 an 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 an 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 an 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 an ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established' when the user is added to the conference.
Configuration	ISUP/BICC interworking applies in Network A User in Network A is subscribed to the 3PTY supplementary service
SIP Parameter	INFO1 <b1> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required</b1>
	CPG Generic Notification conference established
	INFO2 <b1> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic Notification Other party added</b1>
	INFO3 <b2> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic Notification</b2>
Message flow SIP (Network A) Establish a conf	conference established Interconnection Interface SIP (Network B) firmed 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)
Establish a confirmed e	← 200 INFO
	<pre>conference INFO2(Call-ID A-B2, CPG) → € 200 INFO</pre>
	INFO3(Call-ID A-B2, CPG) → 200 INFO Apply post test routine

Comments	User A establishes confirmed communication to user B1 in Network B and invoke the CONF communication.
	Check: Is an INFO request sent to user B1 and in Network B and Is an 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 an 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 an 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 [Network A] SE 47 AND SE 56		
Test purpose	SIP-I/ISUP interworking. Isolation and Reattachment of one party of the		
	conference.		
	Served User A is located in Network A and ISUP/BICC - SIP-I interworking applies in Network A. User A invokes a CONF communication with user B1 and user B2 in Network B. Ensure that when User A isolates one remote party (B1) from the CONF communication:		
	 an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'isolated' in the encapsulated ISUP/BICCCPG; an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party isolated' in the encapsulated ISUP/BICCCPG. Ensure that when User A reattaches one remote party (B1) to the CONF 		
	communication:		
	 an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'reattached' in the encapsulated ISUP/BICCCPG; an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party reattached' in the encapsulated ISUP/BICCCPG. 		
Configuration	ISUP/BICC interworking applies in Network A		
	User in Network A is subscribed to the 3PTY supplementary service		
SIP Parameter	INFO1 <b1> CPG Generic Notification= isolated</b1>		
	INFO2 <b1> CPG Generic Notification= Other party isolated</b1>		
	INFO3 <b2> CPG Generic Notification= reattached</b2>		
	INFO4 <b2> CPG Generic Notification= Other party reattached</b2>		

Message flow					
SIP (Network A)		Interconnection Interface	SIP (Network B)		
Establish a CONF communication with User B1 and User B2 in Network B					
ί ί	Jser A isola	tes User B1 from the CONF conver	sation		
		INFO1(Call-ID A-B1, CPG)	→		
	+	200 INFO			
		INFO3(Call-ID A-B2, CPG)	→		
	÷	200 INFO			
L	Jser A reatt	ach <mark>es Us</mark> er B1 to the CONF conver	sation		
		INFO2(Call-ID A-B2, CPG)	→		
	+	200 INFO			
		INFO4(Call-ID A-B2, CPG)	→		
	÷	200 INFO			
		Apply post test routine			
Comments		nvokes a CONF conversation with Us			
		splits user B1 in Network B from the C			
	Check:	Is an INFO request sent to user B1			
		to 'isolated' in the encapsulated CPC			
	Check:	Is an INFO request sent to user B2			
		to 'Other party isolated' in the encap			
		eattaches user B1 in Network B to the			
	Check:	Is an INFO request sent to user B1			
		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.			

Test case number	SS conf 008
	SIP-SIP/Service/CONF
Test case group Reference	
SELECTION EXPRESSION	5.4/[24]
	[Network A] SE 17 AND [Network A] 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
	INFO3 <b2></b2>
	CPG
	Generic Notification=Conference established
	INFO4 <b2></b2>
	CPG
	Generic Notification= Other party added
Message flow	Generic Notification= Other party added
SIP (Network A)	Interconnection Interface SIP (Network B)
	CONF communication with User B1 and User B2 in Network B ser A isolates User B1 from the CONF conversation
	INFO1(Call-ID A-B1, CPG) →
	← 200 INFO
1	INFO3(Call-ID A-B2, CPG) →
	← 200 INFO
Us	er A reattaches User B1 to the CONF conversation INFO2(Call-ID A-B2, CPG) →
	← 200 INFO
	INFO4(Call-ID A-B2, CPG) → 200 INFO
	Apply post test routine
Comments	User A Invokes a CONF conversation with User B1 and User b2 in Network B.
Comments	User A invokes a CONF conversation with User B1 and User b2 in Network B.
	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?
1	Check: Is an INFO request sent to user B2 and the Generic notification is set
	to 'Other party added' in the encapsulated CPG? Repeat this test in reverse direction.

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

Test case number	SS_acr-cb_00	1		
Test case group	SIP-SIP/Service/ACR-CB			
Reference	4.5.2.6/ [12]			
SELECTION EXPRESSION	SE 32			
Test purpose	Call Barring p	performed in network B fo	or user B.	
		ted in network A and user E the Incoming Call Barring s		n network B and is
	Ensure that a d	communication from user A	is rejected i	n network B by sending a
	603 Decline du	ue to the Call Barring servic	e of user B.	
Configuration	User B is subs	cribed to the incoming Call	Barring serv	vice (e.g. user A in a black
_	list)			
SIP Parameter	INVITE			
	P-A	sserted-Identity: <uri of="" th="" us<=""><th>ser A></th><th></th></uri>	ser A>	
Message flow SIP (Network A)	Int	erconnection Interface		SIP (Network B)
		INVITE	→	
	+	603 (Decline)	-	
	-	ACK	→	
Comments	Check: Is th	ne P-Asserted-Identity pres	ent?	
	Check: Is th	ne communication rejected	by sending a	a 603 (Decline) final
		ponse sent to user A?		
		st in reverse direction.		

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

Instruction Image: Second Silp-Silp/Service/ACR-CB Reference 6.5/[24] SELECTION EXPRESSION [Network B] SE 17 AND SE 47 AND SE 57 Test purpose SIP-1 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. An ISUP/BICC REL is present in the 603 the Cause indicator value is set to '21' if SIP-1 - 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-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause indicator ACK Comments Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id? Check: Is the Privacy header set to 'id? Check: Is the Privacy header set to 'id? Check: Is an ISUP/BICC REL is present in the 603 and the cause value is set</uri>	Test case number	SS acr-cb 003		
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. An ISUP/BICC REL is present in the 603 the Cause indicator value is set to '21' if SIP-I - ISUP/BICC interworking is applicable in Network B. Configuration User B is subscribed to the Anonymous Call Rejection service SIP Parameter INVITE P-Asserted-Identity: <uri a="" of="" user=""> Privacy: id 433 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause indicator 2 Contents Interconnection Interface SIP (Network B) INVITE → € 603 Decline (REL) ACK ACK → Comments Check: Is the Privacy header set to 'id'? Check: Is the Privacy header set to 'id'? Check:</uri>				
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. An ISUP/BICC REL is present in the 603 the Cause indicator value is set to '21' if SIP-1 - 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: Is the P-Asserted-Identity resent?</uri>	v 1			
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. An ISUP/BICC REL is present in the 603 the Cause indicator value is set to '21' if SIP-I - ISUP/BICC interworking is applicable in Network B. Configuration User B is subscribed to the Anonymous Call Rejection service SIP Parameter INVITE P-Asserted-Identity: <uri a="" of="" user=""> Privacy: id 433 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause = 21 Message flow SIP (Network A) Interconnection Interface ACK SIP (Network B) Comments Check: Is the P-Asserted-Identity present? Check: Check: Is the Privacy header set to 'id'? Check: Check: Is the Privacy header set to 'id'? Check: Is the Privacy header set to 'id'? Check: Is an ISUP/BICC REL is present in the 603 and the cause value is set</uri>				
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. An ISUP/BICC REL is present in the 603 the Cause indicator value is set to '21' if SIP-1 - 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-Type: application/isup;version=itu-t92 Content-Type: application Interface SIP (Network B) INVITE → Cause indicator Cause = 21 Message flow SIP (Network A) SIP (Network A) Interconnection Interface SIP (Network B) INVITE → € 603 Decline (REL) → ACK → Comments Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Comments Check: Is the communicat</uri>				
indicator value is set to '21' if SIP-I - ISUP/BICC interworking is applicable in Network B. Configuration User B is subscribed to the Anonymous Call Rejection service SIP Parameter INVITE P-Asserted-Identity: <uri a="" of="" user=""> Privacy: id 433 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause = 21 Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → € 603 Decline (REL) ACK → Comments Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the Privacy bending 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</uri>		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		
Network B. Configuration User B is subscribed to the Anonymous Call Rejection service SIP Parameter INVITE P-Asserted-Identity: <uri a="" of="" user=""> Privacy: id 433 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause indicator SIP (Network A) Interconnection Interface SIP (Network B) INVITE → 603 Decline (REL) → ACK → Comments Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the Privacy header set to yearding 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</uri>				
Configuration User B is subscribed to the Anonymous Call Rejection service SIP Parameter INVITE P-Asserted-Identity: <uri a="" of="" user=""> Privacy: id 433 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause = 21 Message flow SIP (Network A) Interconnection Interface INVITE Comments Check: Is the P-Asserted-Identity present? Check: Is the P-Asserted-Identity present? Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the communication rejected by sending a 603 Decline final response sent to user A? Check: Is an ISUP/BICC REL is present in the 603 and the cause value is set</uri>		indicator value is set to '21' if SIP-I - ISUP/BICC interworking is applicable in		
SIP Parameter INVITE P-Asserted-Identity: <uri a="" of="" user=""> Privacy: id 433 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause = 21 REL: Cause = 21 Message flow SIP (Network A) Interconnection Interface ACK SIP (Network B) INVITE VITE ACK → Comments Check: Is the P-Asserted-Identity present? Check: Check: Is the Privacy header set to 'id'? Check: Check: Is the communication rejected by sending a 603 Decline final response sent to user A? Check:</br></br></br></uri>				
P-Asserted-Identity: <uri a="" of="" user=""> Privacy: id 433 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause = 21 Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → € 603 Decline (REL) ACK → Comments Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the 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</uri>				
Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause = 21 Message flow SIP (Network A) Interconnection Interface SIP (Network A) Interconnection Interface SIP (Network B) INVITE ACK ACK SIP (Network B) Inversion (REL) ACK Comments Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the communication rejected by sending a 603 Decline final response sent to user A? Check: Is an ISUP/BICC REL is present in the 603 and the cause value is set	SIP Parameter	P-Asserted-Identity: <uri a="" of="" user=""></uri>		
Message flow SIP (Network A) Interconnection Interface INVITE SIP (Network B) ← 603 Decline (REL) ACK → Comments Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the communication rejected by sending a 603 Decline final response sent to user A? Check: Is an ISUP/BICC REL is present in the 603 and the cause value is set		Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator		
SIP (Network A) Interconnection Interface SIP (Network B) INVITE → 603 Decline (REL) ACK → ACK → Comments Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the communication rejected by sending a 603 Decline final response sent to user A? Check: Is an ISUP/BICC REL is present in the 603 and the cause value is set	Message flow	04036 - 21		
Comments Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the communication rejected by sending a 603 Decline final response sent to user A? Check: Is an ISUP/BICC REL is present in the 603 and the cause value is set		INVITE → € 603 Decline (REL)		
 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 				
 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 	Comments			
response sent to user A? Check: Is an ISUP/BICC REL is present in the 603 and the cause value is set				
Check: Is an ISUP/BICC REL is present in the 603 and the cause value is set				
10 21 :		Check: Is an ISUP/BICC REL is present in the 603 and the cause value is set		
Repeat this test in reverse direction.				

Test case number	SS_cug_	001		
Test case group		Service/CUG		
Reference		4.5.2.4/ [13]		
SELECTION EXPRESSION	SE 34			
Test purpose		ng user +OA to terminating user n		
		ating user in a CUG Outgoing Access e session establishment is successful		
Configuration	Originati	ing user: CUG, outgoing access allow	ved	
SIP Parameter	INVITE:			
	Conte	ent-Type: application/vnd.etsi.cug+xm	l	
	Conte	ent-Disposition: signal;handling=		
	 <cl< th=""><th></th><th></th></cl<>			
		networkIndicator>01 networkInd</th <th>dicator</th>	dicator	
		networkIndicator>23 networkInd</th <th></th>		
		cugInterlockBinaryCode>0F03 </th <th></th>		
		<cugcommunicationindicator>10</cugcommunicationindicator>		
Message flow	<0	ug>		
SIP (Network A)		Interconnection Interface	SIP (Network B)	
		INVITE		
	←	180 Ringing	2	
		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 INVIT		
	Check:	Contains the XML body in the INVIT	E a 'networkIndicator' element as	
		a 'cug' child element?		
	Check:	Contains the XML body in the INVIT	FE a 'cugInterlockBinaryCode'	
		element as a 'cug' child element?		
	Check:	Contains the XML body in the INVIT		
	Cheeler	element set to '10' as a 'cug' child e		
	Check:	Is the session setup not rejected?		
	NOTE:	his test in reverse direction.	and the augletoria de Dinam Carla	
	NOTE:	The networkIndicator element value	and the cuginteriockBinaryCode	
		element value are examples.		

7.1.5.9 Closed User Group (CUG)

Test case number	SS_cug_	002		
Test case group	SIP-SIP/Service/CUG			
Reference	4.5.2.4, 4.5.2.10/ [13]			
SELECTION EXPRESSION	SE 34			
Test purpose		ing user -OA to terminating user no CUG.		
	onginat	Originating user -OA to terminating user no 606.		
		An originating user in a CUG Outgoing Access not allowed calls a user not in a CUG. The session establishment is not successful, a 403 (Forbidden) response is sent		
Configuration	Originat	ing user: CUG, outgoing access not allowed		
SIP Parameter	INVITE:			
		ent-Type: application/vnd.etsi.cug+xml		
		ent-Disposition: signal;handling= required		
	<:C	ug>		
	<	networkIndicator>01		
	<.	networkIndicator>23		
	<.	ugInterlockBinaryCode>0F03		
		cugCommunicationIndicator>11		
	<cl< th=""><th colspan="3"><cug></cug></th></cl<>	<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'		
		element as a 'cug' child element?		
	Check:	Contains the XML body in the INVITE a 'cugCommunicationIndicator'		
		element set to '11' as a 'cug' child element?		
	Check:	Is the session setup rejected? A 403 (Forbidden) final response is sent		
		by the terminating network?		
		his test in reverse direction.		
	NOTE:	The networkIndicator element value and the cugInterlockBinaryCode		
		element value are examples.		

Test case number	SS_cug	003	
Test case group	SIP-SIP/Service/CUG		
Reference	4.5.2.4, 4.5.2.10/ [13]		
SELECTION EXPRESSION			
	SE 34		
Test purpose	An origin	ing user -OA to terminating user -IA. ating user in a CUG Outgoing Access not allowed calls a user in the	
	same CUG Incoming Access not allowed. The session establishment is successful.		
O and investigation			
Configuration		ing user: CUG, outgoing access not allowed	
		ting user: CUG incoming access not allowed	
		etwork A and user in network B are in the same CUG	
SIP Parameter	INVITE:		
		ent-Type: application/vnd.etsi.cug+xml	
	Conte	ent-Disposition: signal;handling= required	
	<cl< th=""><th>ıg></th></cl<>	ıg>	
	<.	networkIndicator>01	
	<.	networkIndicator>23	
	<cuginterlockbinarycode>0F03</cuginterlockbinarycode>		
	<cug></cug>		
Message flow	•	x	
SIP (Network A)		Interconnection Interface SIP (Network B)	
		INVITE ->	
	←	180 Ringing	
		Apply post test routine	
Comments	Check:	Is the Content-Type in The INVITE set to	
		application/vnd.etsi.cug+xml?	
	Check:	Is the handling parameter in the Content-Disposition header set to	
		required?	
	Check:	Contains the XML body in the INVITE a 'cug' element?	
	Check:	Contains the XML body in the INVITE a 'networkIndicator' element as	
	oneon.	a 'cug' child element?	
		Contains the XML body in the INVITE a 'cugInterlockBinaryCode'	
	Check.		
	Check:		
		element as a 'cug' child element?	
	Check: Check:	element as a 'cug' child element? Contains the XML body in the INVITE a 'cugCommunicationIndicator'	
	Check:	element as a 'cug' child element? Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element?	
	Check: Check:	element as a 'cug' child element? Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? Is the session setup not rejected?	
	Check: Check: Repeat ti	element as a 'cug' child element? Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? Is the session setup not rejected? his test in reverse direction.	
	Check: Check:	element as a 'cug' child element? Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? Is the session setup not rejected?	

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.		
	not allow response			
Configuration		network A and user in network B are not in the same CUG		
		ting user: CUG incoming access not allowed		
SIP Parameter	INVITE:			
		ent-Type: application/vnd.etsi.cug+xml		
	Conte	ent-Disposition: signal;handling=		
	<cl< th=""><th>5</th></cl<>	5		
		networkIndicator>01		
		networkIndicator>23		
		<cuginterlockbinarycode>0F03</cuginterlockbinarycode>		
		<cugcommunicationindicator></cugcommunicationindicator>		
	<cl< th=""><th>ug></th></cl<>	ug>		
Message flow SIP (Network A)		Interconnection Interface SIP (Network B)		
	←	403 (Forbidden)		
	•	ACK →		
Comments	Check:	Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml?		
	Check:	Contains the XML body in the INVITE a 'cug' element?		
	Check:	Contains the XML body in the INVITE a 'networkIndicator' element as		
		a 'cug' child element?		
	Check:	Contains the XML body in the INVITE a 'cugInterlockBinaryCode'		
		element as a 'cug' child element?		
	Check:	Contains the XML body in the INVITE a 'cugCommunicationIndicator'		
		element set to '10' or '11'as a 'cug' child element?		
	Check:	Is the session setup rejected? A 403 (Forbidden) final response is sent		
		by the terminating network?		
	Repeat t	his test in reverse direction.		
	NOTE:	The networkIndicator element value and the cugInterlockBinaryCode		
		element value are examples.		

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

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

Test case number	SS_cug_	007
Test case group		Service/CUG
Reference	4.5.2.4/ [
SELECTION EXPRESSION	SE 34	
Test purpose		ing user -OA, network B does not support CUG.
	e i ginat	
	An origin	ating user in a CUG Outgoing Access not allowed calls a user in
		B. Network B does not support CUG. The session establishment is not
		ul, a 4xx unsuccessful final response is sent.
Configuration		
SIP Parameter	INVITE:	
	Conte	ent-Type: application/vnd.etsi.cug+xml
	Conte	ent-Disposition: signal;handling= required
	<cl< th=""><th>0</th></cl<>	0
		networkIndicator>01
		networkIndicator>23
		cugInterlockBinaryCode>0F03
		cugCommunicationIndicator>11
	<cl< th=""><th>ug></th></cl<>	ug>
Message flow		Interconnection Interface SID (Network P)
SIP (Network A)		Interconnection Interface SIP (Network B) INVITE →
	←	4xx/501 Not Implemented
	•	ACK →
Comments	Check:	Is the Content-Type in The INVITE set to
	•	application/vnd.etsi.cug+xml?
	Check:	Is the handling parameter in the Content-Disposition header set to
		required?
	Check:	Contains the XML body in the INVITE a 'cug' element?
	Check:	Contains the XML body in the INVITE a 'networkIndicator' element as
		a 'cug' child element?
	Check:	Contains the XML body in the INVITE a 'cugInterlockBinaryCode'
		element as a 'cug' child element?
	Check:	Contains the XML body in the INVITE a 'cugCommunicationIndicator'
		element set to '11' as a 'cug' child element?
	Check:	Is the session setup rejected by sending an unsuccessful final
	Denest	response?
		his test in reverse direction.
	NOTE:	The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug	007A	
Test case group		Service/CUG	
Reference	4.5.2.4/ [
SELECTION EXPRESSION	SE 34		
Test purpose		Originating user CUG-OA to terminating CUG user +ICB	
	Originat		
	An origin	ating user in a CUG outgoing access not allowed calls a user in the	
		IG Incoming communication barred. The session establishment is not	
		ul, a 603 (Decline) response is sent.	
Configuration		letwork B in a CUG incoming Communication Barring	
SIP Parameter	INVITE:		
		ent-Type: application/vnd.etsi.cug+xml	
		ent-Disposition: signal;handling= required	
	<cl< th=""><th><0></th></cl<>	<0>	
		networkIndicator>01	
		networkIndicator>23	
		cugInterlockBinaryCode>0F03	
		cugCommunicationIndicator>11	
	<cl< th=""><th></th></cl<>		
Message flow		.2	
SIP (Network A)		Interconnection Interface SIP (Network B)	
		INVITE ->	
	←	603 Decline	
		ACK →	
Comments	Check:	Is the Content-Type in The INVITE set to	
		application/vnd.etsi.cug+xml?	
	Check:	Is the handling parameter in the Content-Disposition header set to	
		required?	
	Check:	Contains the XML body in the INVITE a 'cug' element?	
	Check:	Contains the XML body in the INVITE a 'networkIndicator' element as	
		a 'cug' child element?	
	Check:	Contains the XML body in the INVITE a 'cugInterlockBinaryCode'	
		element as a 'cug' child element?	
	Check:	Contains the XML body in the INVITE a 'cugCommunicationIndicator'	
		element set to '11' as a 'cug' child element?	
	Check:	Is the session setup rejected by sending a 603 Decline unsuccessful	
		final response?	
	Repeat t	his test in reverse direction.	
	NOTE:	The networkIndicator element value and the cugInterlockBinaryCode	
		element value are examples.	

Test case number	SS_cug_008		
Test case group	SIP-SIP/Service/CUG		
Reference	7.1/[24]		
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] 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 an 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)		
	← 180 Ringing		
	Apply post test routine		
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 [Network A] SE 47 AND SE 58
Test purpose	SIP-I/ISUP interworking. CUG call with outgoing access not allowed (both
Test purpose	user in different CUG).
	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 unsuccessful due to both CUG user are in different CUG. There is an 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
Maaaaaa flaw	[any boundary name]
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →
	← 500 Server Internal error ACK →
	Apply post test routine
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 [Network A] SE 47 AND SE 58) AND ([Network B] SE	F
	17 AND [Network B] SE 47 AND SE 58)	
Test purpose	SIP-I/ISUP interworking. CUG call with outgoing access not allowed (both	h
rest purpose	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 a SIP-I - ISUP/BICC interworking applies in the same CUG. Ensure that when us A is in a CUG 'outgoing access not allowed' calls user B in Network B. The call successful. There is an Optional forward call indicator the CUG Call Indicator Outgoing access not allowed present in the encapsulated IAM sent to Network B.	ser
Configuration	User in PSTN/PLMN part of Network A in a CUG outgoing access not	t
-	allowed	
	 User in PSTN/PLMN part of Network B in a CUG 	
	User A and User B are in the same CUG	
SIP Parameter	INVITE	
	Content-Type: multipart/mixed;boundary=[any boundary name]	
	[any boundary name]	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	IAM	
	Optional Forward call indicator	
	CUG Call Indicator	
	Outgoing access not allowed	
	CUG interlock code	
	[any boundary name]	
Message flow		
SIP (Network A)	Interconnection Interface SIP (Network B)	
	INVITE →	
	← 180 Ringing	
	Apply post test routine	
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	το
	Network B? Check: Is the Optional forward call indicator present, the CUG Call Indicator	r ic
	set to 'Outgoing access not allowed'?	1 15
	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 [Network A] SE 47 AND SE 58) AND ([Network B] SE
	17 AND [Network B] 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 use A is in a CUG 'outgoing access not allowed' calls CUG user B in Network B. The call is successful. There is an Optional forward call indicator the CUG Call Indicator Outgoing access not allowed present in the encapsulated IAM sent to Network B.
Configuration	 User in PSTN/PLMN part of Network A in a CUG outgoing access not allowed
	User in PSTN/PLMN part of Network B in a CUG incoming access not allowed
	User A and User B are in the same CUG
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required
	IAM Optional Forward call indicator
	CUG Call Indicator
	Outgoing access not allowed
	CUG interlock code
	[any boundary name]
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →
	← 180 Ringing
	Apply post test routine
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 [Network A] SE 47 AND SE 58) AND ([Network B] SE 17 AND [Network B] 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 an 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. An ISUP/BICC		
	REL is encapsulated and the Cause value is set to '87'.		
Configuration	 User in PSTN/PLMN part of Network A in a CUG outgoing access not allowed 		
	 User in PSTN/PLMN part of Network B in a CUG incoming access not allowed 		
	User A and User B are in different CUG		
SIP Parameter	INVITE		
	Content-Type: multipart/mixed;boundary=[any boundary name]		
	[any boundary name]		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	IAM		
	Optional Forward call indicator CUG Call Indicator Outgoing access not allowed CUG interlock code		
	[any boundary name]		
	500		
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	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) ACK →		
Comments	User A in the PSTN/PLMN part of Network A calls user B in Network B.		
	User B in the PSTN/PLMN part of Network B.		
	Check: Is an IAM encapsulated in the INVITE request sent from Network A to Network B?		
	Check: Is the Optional forward call indicator present, the CUG Call Indicator is		
	set to 'Outgoing access not allowed'?		
	Check: Is the CUG interlock code parameter present in the encapsulated		
	IAM? Check: Is the call rejected with a 500 final response and an 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 and not in a CUG. User B is in a CUG Incoming access not allowed and 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. An ISUP/BICC REL is encapsulated and the Cause value is set to '87'.		
Configuration	User in PSTN/PLMN part of Network B in a CUG incoming access not allowed		
SIP Parameter	500 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause indicators Cause value 87		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → € 500 Server Internal error(REL) ACK →		
Comments	 User A in Network A calls user B in Network B. User B in the PSTN/PLMN part of Network B. Check: Is the call rejected with a 500 final response and an ISUP/BICC REL is encapsulated and the cause value is set to 87? Repeat this test in reverse direction. 		

Test sees number	
Test case number	SS_cug_014
Test case group	SIP-SIP/Service/CUG
Reference	7.1/[24]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 58
Test purpose	SIP-I/ISUP interworking. Call to a CUG user incoming access allowed.
	User A is located in Network A and not in a CUG. User B is located in the PSTN/PLMN part and SIP-I - ISUP/BICC interworking applied. Ensure that when user A calls CUG user B Incoming access allowed in Network B. The call is successful.
Configuration	 User in PSTN/PLMN part of Network B in a CUG incoming access allowed
SIP Parameter	
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → € 180 Ringing
	Apply post test routine
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.

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.	

199

Test case number	SS_cw_0	002		
Test case group	SIP-SIP/S	Service/CW		
Reference	4.5.5.2/ [15]		
SELECTION EXPRESSION	SE 35 AN	ND SE 36		
Test purpose	Call reje	cted after timeout TAS-CW.		
	the comn Ensure th Waiting ir or UDUB	located in network A, user B is lo nunication Waiting service. nat when user A calls user B, user ndication' in the 180 Ringing provi . After timeout TAS-CW network B ble) response toward user A and th	· A receives sional resp 3 sends a 4	s the 'communication onse if the user B is NDUB I80 (Temporarily
Configuration				
SIP Parameter	180: Alert- 480:	Info: <urn:alert:service:call-waiting< th=""><th>]></th><th></th></urn:alert:service:call-waiting<>]>	
	Rease	on: Q.850 ;cause=19		
Message flow SIP (Network A)		Interconnection Interface INVITE	→	SIP (Network B)
	+	180 Ringing Timeout TAS-CW		
	+	480 (Temporarily unavailable) ACK	→	
Comments	Check:	Is an Alert-Info header present in value set to ' <ur></ur>		
	Check:	Is a Reason header present in the set to 'Q.850' and the cause para		
	Repeat th	nis test in reverse direction.		

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

Test case number	SS_cw_004
Test case group	SIP-SIP/Service/CW
Reference	6.5/ [24]
SELECTION EXPRESSION	[Network B] SE 17 AND [Network B] SE 47 AND SE 59
Test purpose	SIP-I support. Call Waiting indication in 180 with encapsulated CPG.
rest purpose	on a support. Oan waiting indication in 100 with encapsulated of 0.
	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
Manager flam	call is a waiting call
Message flow	Interconnection Interface SIP (Network B)
SIP (Network A)	Interconnection Interface SIP (Network B)
	← 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 call'?
	Repeat this test in reverse direction.

Test case number	SS_ect_001	1	
Test case group	SIP-SIP/Ser		
Reference	4.5.2/[11]		
SELECTION EXPRESSION		SE 37 AND [Network A] SE	11 AND [Network A] SE 49
Test purpose		red transfer using the REF	
	User A is loo A invokes E • Ens dial ado 'IN\ • Ens the	cated in network A, user B a CT to transfer a session with sure that a REFER request is logue with user B. The URI in dress of the ECT AS in netwo VITE'. sure that an INVITE request Request URI is set to the ac	nd user C are located in network B. User
		Request URI is set to the ac	
Configuration		Trequest OTT is set to tile at	
SIP Parameter		quest URI address of user B	
		o: <uri ect-as="" of="">; method</uri>	
	INVITE1 Re	equest URI address of ECT-A	\S
Message flow	INVITE2: Re	equest URI address of user (C
A confi	rmed session	n is established between us n is established between us ECT to transfer the session REFER 202 Accepted NOTIFY (100) 200 OK NOTIFY	ser A and user C
CA	SE Blind tran		
	+	BYE (A-B) 200 OK BYE	→
	•	200 OK DIL	
	÷	INVITE1 (ECT-AS)	
	-	INVITE2 (user C)	→
	← ←	INVITE2 (user C) 200 OK INVITE	→
	-	INVITE2 (user C) 200 OK INVITE ACK	→
	(INVITE2 (user C) 200 OK INVITE ACK 200 OK INVITE	→ → →
	-	INVITE2 (user C) 200 OK INVITE ACK 200 OK INVITE ACK	→
	«	INVITE2 (user C) 200 OK INVITE ACK 200 OK INVITE ACK NOTIFY (200)	→
	«	INVITE2 (user C) 200 OK INVITE ACK 200 OK INVITE ACK	→→
CAS	«	INVITE2 (user C) 200 OK INVITE ACK 200 OK INVITE ACK NOTIFY (200) 200 OK NOTIFY	→→
CAS	← ← ← SE Assured tra	INVITE2 (user C) 200 OK INVITE ACK 200 OK INVITE ACK NOTIFY (200) 200 OK NOTIFY ansfer BYE (A-B)	→→
CAS	← ← Æ SE Assured tra ←	INVITE2 (user C) 200 OK INVITE ACK 200 OK INVITE ACK NOTIFY (200) 200 OK NOTIFY	 → → →

7.1.5.11 Explicit Communication Transfer (ECT)

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:	
		address of the ECT-AS in network A?
	Check:	Is an INVITE request is sent to network B the Request is set to the
	C	address of user C?
	Check:	When the session from user B to user C is confirmed a NOTIFY
	encon	request is sent to network A containing sipfrag body set to 'SIP/2.0
		200 OK' and if Assured transfer is applicable the session from user A
		to user B is terminated by user A.
	Check:	
	Repeat the	his test in reverse direction.

Test case number	SS_ect_002
Test case group	SIP-SIP/Service/ECT
Reference	4.5.2/ [11]
SELECTION EXPRESSION	[Network A] SE 37 AND [Network A] SE 11 AND [Network A] SE 50
Test purpose	Consultative transfer using the REFER method.
Configuration SIP Parameter	 User A is located in network A, user B and user C are located in network B. User A invokes ECT to transfer a session with user B to user C. Ensure that a REFER request is sent from network A to network B in the dialogue with user B. The URI in the Refer-To header is set to the address of the ECT AS in network A and the method parameter is set to "INVITE'. Ensure that an INVITE request is sent from network B to network A and the Request URI is set to the address of the ECT AS in network A. Ensure that an INVITE request is sent from network A to network B and the Request URI is set to the address of user C and a Replaces header is present containing the session identifiers of the session A - C. REFER :Request URI address of user B Refer-To: <uri ect-as="" of="">; method=invite</uri>
Message flow	INVITE2: Request URI address of user C Require: replaces Replaces: <session a-c=""></session>
A confir	med session is established between user A and user B med session is established between user A and user C er A invokes ECT to transfer the session to user C REFER → 202 Accepted
	 NOTIFY (100) 200 OK NOTIFY →
	 ► INVITE1 (ECT-AS) INVITE2 (user C) → € 200 OK INVITE ACK →
	BYE (A-B) → 200 OK BYE
	← BYE (A-C) 200 OK BYE → Apply post test routine
Comments	 Check: Is a REFER request is sent network B, the Refer-To header is set to the URI of the ECT-AS in network A and a method parameter is present set to 'INVITE'? Check: Is an INVITE request sent to network A the Request line is set to the
	 Check: Is an INVITE request sent to network A the Request line is set to the address of the ECT-AS in network A? Check: Is an INVITE request is sent to network B the Request is set to the address of user C and a Replaces header is present contains the session identifiers of the session A-C?
	Check: Is the session A - B and the session A - C terminated?Check: Ensure the property of speech between user B and user C.Repeat this test in reverse direction.

Test case number	SS_ect_003
Test case group	SIP-SIP/Service/ECT
Reference	4.5.2/ [11], 4.7.2.9.7/ [20]
SELECTION EXPRESSION	[Network A] SE37 AND [Network A] SE 12 AND [Network A] SE 49
Test purpose	Blind/assured transfer using the 3pcc method.
	User A is located in network A, user B and user C are located in network B User
	A invokes ECT to transfer a session with user B to user C.
	 Ensure that the network A establishes a session to user C.
	 Ensure that the network A sends a reINVITE to update the session
	between user A and user B (SDP: IP address, port and codec).
Configuration	
SIP Parameter	INVITE1 Request URI address of user C
	CASE A
	INVITE2: Request URI address of user B
	SDP
	c=IN IP4/6 [new IP address]
	m=audio [new port] RTP/AVP [new codec list]
	a=[new attributes]
	CASE B
	INVITE2: Request URI address of user B
	SDP
	a=[new attributes]
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	med session is established between user A and user B
	er A invokes ECT to transfer the session to user C
	INVITE1 (user C) →
	← 180 Ringing
	← 200 OK INVITE
	ACK 7
	INVITE2 (user B) →
	← 200 OK INVITE
Commonto	Apply post test routine
Comments	Check: Is an initial INVITE is sent from network A to user C to establish a
	dialogue between network A and user C?
	Check: Is a reINVITE is sent from network A to user B update the session
	parameter in the SDP? CASE A reflects the ECT procedure in the
	IMS, CASE B reflects the interworking in the MGCF.
1	Repeat this test in reverse direction.

Test case number SS_ect_004 test case group SIP-SIP/Service/ECT teference 4.5.2/ [11], 4.7.2.9.7/ [20] EELECTION EXPRESSION [Network A] SE37 AND [Network A] SE 12 AND [Network A] SE 50 Test purpose Consultative transfer using the 3pcc method. User A is located in network A, user B and user C are located in network B User A invokes ECT to transfer a session with user B to user C. • Ensure that the network A sends a reINVITE to update the session between user A and user B (SDP: IP address, port and codec). • Ensure that the network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). configuration INVITE1: Request URI address of user C SDP SiP Parameter INVITE1: Request URI address of user C SDP CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE A CASE B CASE B CASE B D CASE B D CASE B D CASE B D
Leference 4.5.2/ [11], 4.7.2.9.7/ [20] IELECTION EXPRESSION [Network A] SE37 AND [Network A] SE 12 AND [Network A] SE 50 Consultative transfer using the 3pcc method. User A is located in network A, user B and user C are located in network B User A invokes ECT to transfer a session with user B to user C. • Ensure that the network A sends a reINVITE to update the session between user A and user B (SDP: IP address, port and codec). • Ensure that the network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). • Ensure that the network A sends a reINVITE to update the session between user A and user C (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). • Ensure that the network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). • Ensure that the network P sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). • Ensure that the network B sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). • Ensure that the network B sends a reINVITE to update the session between user A and user C (SDP: IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE A INVITE2: Request URI address]<
ELECTION EXPRESSION [Network A] SE37 AND [Network A] SE 12 AND [Network A] SE 50 Test purpose Consultative transfer using the 3pcc method. User A is located in network A, user B and user C are located in network B User A invokes ECT to transfer a session with user B to user C. • Ensure that the network A sends a relNVITE to update the session between user A and user B (SDP: IP address, port and codec). • Ensure that the network A sends a relNVITE to update the session between user A and user C (SDP: IP address, port and codec). Configuration 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] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B EASE B
Test purpose Consultative transfer using the 3pcc method. User A is located in network A, user B and user C are located in network B User A invokes ECT to transfer a session with user B to user C. • Ensure that the network A sends a reINVITE to update the session between user A and user B (SDP: IP address, port and codec). • Ensure that the network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). • Ensure that the network A sends a reINVITE to update the session between user A and user C (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). • Ensure that the network A sends a reINVITE to update the session between user A and user C (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). • Ensure that the network A sends a reINVITE to update the session between user A and user C (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 • 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] • CASE B • INVITE2
User A is located in network A, user B and user C are located in network B User A invokes ECT to transfer a session with user B to user C. • Ensure that the network A sends a reINVITE to update the session between user A and user B (SDP: IP address, port and codec). • Ensure that the network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). • Ensure that the network A sends a reINVITE to update the session between user A and user C (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). • Ensure that the network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). • Ensure that the network I address of user C SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
User A invokes ECT to transfer a session with user B to user C. Ensure that the network A sends a reINVITE to update the session between user A and user B (SDP: IP address, port and codec). Ensure that the network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). Configuration 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] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
User A invokes ECT to transfer a session with user B to user C. Ensure that the network A sends a reINVITE to update the session between user A and user B (SDP: IP address, port and codec). Ensure that the network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). Configuration 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] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
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] CASE A SDP c=IN IP4/6 [new IP address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
Ensure that the network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). 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] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE A INVITE2: Request URI address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
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] CASE A NVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
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] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes]
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] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
a=[new attributes] CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
CASE A INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
m=audio [new port] RTP/AVP [new codec list] a=[new attributes] CASE B
a=[new attributes]
CASE B
INVITE2: Request URI address of user B
SDP
a=[new attributes]
lessage flow
SIP (Network A) Interconnection Interface SIP (Network B)
A confirmed session is established between user A and user B
A confirmed session is established between user A and user C
User A invokes ECT to transfer the session to user C
INVITE1 (user B) →
← 200 OK INVITE
ACK →
INVITE2 (user C) →
← 200 OK INVITE
Apply post test routine
Comments Check: Is a reINVITE is sent from network A to user B update the session
parameter in the SDP?
Check: Is a reINVITE is sent from network A to user C update the session
parameter in the SDP? CASE A reflects the ECT procedure in the
IMS, CASE B reflects the interworking in the MGCF.
Repeat this test in reverse direction.

Test case number	SS_ect_0	005	
Test case group	SIP-SIP/Service/ECT		
Reference	5.4.3.2/ [24]		
SELECTION EXPRESSION		A] SE 17 AND [Network A] SE 47 AND SE 60	
Test purpose		oport. Call Transfer invoked in active state, call was previous on	
Test purpose	HOLD. BICC/ISU	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.	
		hat a 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			
	Co	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	
	FA	AC	
		Generic Notification	
		Call transfer active	
	-	Call transfer number	
	[a	any boundary name]	
Message flow			
		Interconnection Interface SIP (Network B) stablished between user A and user B and set on hold tes ECT to transfer the session to user C	
		$INFO (LOP request) \rightarrow$	
	←	200 OK INFO	
	÷	INFO (LOP response)	
		200 OK INFO	
CASE A			
		INVITE (sendrecv; FAC) →	
	←	200 OK INVITE	
		ACK →	
CASE B			
		INFO (FAC) →	
	←	200 OK INFO	
		INVITE (sendrecv) →	
	←	200 OK INVITE	
		ACK →	
		Apply post test routine	
Comments	A session	n from User A to User B is already established.	
	User A se	ets the User B on hold.	
	User A in	vokes the ECT service.	
	Check:	Is (optional) an INFO request is sent from Network A to Network B	
		and an ISUP LOP message is present the Loop prevention indicator	
		set to 'request'?	
	Check:	Is (optional) an INFO request is sent from Network A to Network B	
		and an ISUP LOP message is present the Loop prevention indicator	
		set to 'response'?	
	Check:	Is (CASE A) an INVITE request sent and an ISUP FAC message is	
		present containing a Generic notification indicator is set to 'Call	
		transfer active' and in addition the media stream is set to	
		'sendrecv'?	
	Check:	Is (CASE B) an INFO request sent and an ISUP FAC message is	
		present containing a Generic notification indicator is set to 'Call	
		transfer active'? In addition is an INVITE request sent and the media	
		stream is set to 'sendrecv' to resume the held session?	
	NOTE:	The content of the FAC in the INVITE request is Equal to the content	
		of the FAC in the INFO request.	
	Repeat th	nis test in reverse direction.	

Test case number	SS_ect_0	06		
Test case group		Service/ECT		
Reference	5.4.3.2/ [24]			
SELECTION EXPRESSION		[Network A] SE 17 AND [Network A] SE 47 AND SE 60		
			n alerting state, call was previous on	
Test purpose	HOLD. BICC/ISU are locate	IP - SIP-I interworking applies i ed in network A and user B is lo	in the originating network User A and C ocated in network B.	
	and trans	fer the call with User B to User		
Configuration	User A is	subscribed to the Explicit Call	Transfer supplementary service	
SIP Parameter	INVITE Co	ontent-Type: multipart/mixed;bc	oundary=[any boundary name]	
	Co	any boundary name] ontent-Type: application/sdp sendrecv		
	Co			
	[;	Generic Notification Call transfer alerting Call transfer number any boundary name]		
	sion is es	Interconnection Interface tablished between user A an		
Us	er A invok	es ECT to transfer the session		
	←	INFO (LOP request) 200 OK INFO INFO (LOP response)	→	
CASE A	Ľ	200 OK INFO	→	
	÷	INVITE (sendrecv; FAC) 200 OK INVITE ACK	→ →	
CASE B		INFO (FAC)	÷	
	+	200 OK INFO INVITE (sendrecv)	→	
	÷	200 OK INVITE ACK	→	
Comments	Accesion	Apply post test routine from User A to User B is alread	dy astablished	
	User A se A session	ets the User B on hold. I from User A to User C is alreat vokes the ECT service. Is (optional) an INFO reques		
	Check:	set to 'request'? Is (optional) an INFO reques and an ISUP LOP message is	t is sent from Network A to Network B s present the Loop prevention indicator	
	Check:	present containing a Generic	est sent and an ISUP FAC message is notification indicator is set to 'Call	
	Check:	Is (CASE B) an INFO request present containing a Generic transfer alerting'? In addition	on the media stream is set to 'sendrecv'? t sent and an ISUP FAC message is notification indicator is set to 'Call is an INVITE request sent and the media	
	NOTE:	of the FAC in the INFO reque	NVITE request is Equal to the content	
	Repeat th	is test in reverse direction.		

Test case number	SS_ect_00	7	
Test case group		ervice/ECT	
Reference	5.4.3.2/ [24		
SELECTION EXPRESSION] SE 17 AND [Network A] SE 47 AND SE 60	
Test purpose		ort. Call Transfer invoked in active state.	
lest puipose	Sir-i Supp	ont. Can transier invokeu in active state.	
	BICC/ISUE	P - SIP-I interworking applies in the originating network Users A and B	
		I in network A and User C is located in network B.	
		t a User A can successfully invoke the ECT supplementary service	
		er the call with User B to User C in active state.	
Configuration		subscribed to the Explicit Call Transfer supplementary service	
SIP Parameter	INFO	absonce to the Explicit our mansier supplementary service	
	-	ntent-Type: application/isup;version=itu-t92	
		ntent-Disposition: signal;handling=required	
	FAC		
		Generic Notification	
		Call transfer active	
		Call transfer number	
Message flow			
SIP (Network A)		Interconnection Interface SIP (Network B)	
	med sessio	on is established between user A and user C	
	User A invokes ECT to transfer the session to user C		
INFO (LOP request) →			
	←	200 OK INFO	
	←	INFO (LOP response)	
		200 OK INFO 🔶 🔿	
		INFO (FAC) →	
	÷	200 OK INFO	
		Apply post test routine	
Comments		from User A to User B is already established.	
		s the User B on hold.	
	A session	from User A to User C is already established.	
		okes the ECT service.	
		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'?	
		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 '?	
		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	
	NOTE:		

Test case number	SS_ect_0	008
Test case group		Service/ECT
Reference	5.4.3.2/ [2	
SELECTION EXPRESSION		A] SE 17 AND [Network A] SE 47 AND SE 60
Test purpose		port. Call Transfer invoked in alerting state.
	BICC/ISU	IP - SIP-I interworking applies in the originating network User A and B
		ed in network A and user C is located in network B.
		at a User A can successfully invoke the ECT supplementary service
		fer the call with User B to User C in alerting state.
Configuration		subscribed to the Explicit Call Transfer supplementary service
SIP Parameter	INFO	
		ontent-Type: application/isup;version=itu-t92
		ontent-Disposition: signal;handling=required
		PG
		Generic Notification
		Call transfer alerting
		Call transfer number
Message flow		
SIP (Network A)		Interconnection Interface SIP (Network B)
	the early o	lialogue is established between user A and user C
		es 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		from User A to User B is already established.
		ets the User B on hold.
		from User A to User C is already established.
		vokes 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
	NOTT	transfer alerting'?
	NOTE:	The content of the FAC in the INVITE request is Equal to the content
		of the FAC in the INFO request. his test in reverse direction.

7.1.5.12	Malicious Communication	Identification ((MCID))
----------	-------------------------	------------------	--------	---

-		004		
Test case number		SS_mcid_001		
Test case group	SIP-SIP/	SIP-SIP/Service/MCID		
Reference	4.5.2.5/ [4.5.2.5/ [18]		
SELECTION EXPRESSION	SE 38			
Test purpose	Network	Network B sends a MCID request, no response.		
Configuration	the Malic When us request, originatin Ringing r	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.		
SIP Parameter	INFO:	User B is subscribed to the MCID service		
	<.	<:mcid> <:request> <:McidRequestIndicator>01 :McidRequestIndicator <:HoldingIndicator > :HoldingIndicator :request :mcid		
Message flow SIP (Network A)	←	Interconnection Interface INVITE INFO 200 OK INFO Timeout To-ID 180 Ringing Apply post test routine	→ →	SIP (Network B)
Comments	Check:	Is an INFO request sent to net	work A2	
Comments	Check:			0.01'2
	Check:			
		Check: is a 200 OK INFO response sent to network B? Repeat this test in reverse direction.		
	repeari			

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

Test case number	SS_mcid_003			
Test case group	SIP-SIP/Service/MCID			
Reference	5.4.3.2/ [24]			
SELECTION EXPRESSION	([Network B] SE 17 AND SE 47 AND SE 61)			
Test purpose	SIP-I support. Network B sends a MCID request, no response.			
	User A is located in network A, user B is located in the PSTN/PLMN part of network B and subscribed to the Malicious Call Identification service. When user A call user B and no originating identification is present in the INVITE request, the network B sends a 183 Session Progress to network A and an ISUP/BICC IDR message is present the MCID request indicator is set to 'MCID requested' requesting the originating identity. After timeout of timer (ISUP) T39 the network B sends the 180 Ringing response.			
Configuration	User B is subscribed to the MCID service			
SIP Parameter	183: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required			
	MCID request indicators			
	MCID request indicator MCID requested			
Message flow	MCID Tequested			
SIP (Network A)	 Interconnection Interface SIP (Network B) INVITE → 183 Session Progress(IDR) 			
	Timeout T39 ← 180 Ringing(ACM) Apply post test routine			
Comments	Check: Is a 183 Session Progress sent to network A? Check: Is an ISUP/BICC IDR message is present and the MCID request indicator is set to 'MCID requested'?			
	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 A] SE 17 AND SE 47 AND SE 61) AND ([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 a 183 Session Progress to network A requesting the originating identity. After receipt of an INFO request containing the ISUP IRS message from network A the network B sends the 180 Ringing response.		
Configuration	User B subscribed to the MCID service User A is a ISDN or POTS user in the PSTN of network A		
SIP Parameter	183: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	IDR MCID request indicators MCID request indicator MCID requested		
	INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required		
	IRS MCID response indicators MCID response indicator MCID included Calling party number		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE → 183 Session Progress(IDR) INFO(IRS) → 		
	← 200 OK INFO		
	← 180 Ringing(ACM)		
	Apply post test routine		
Comments	Check: Is a 183 Session Progress sent to network A and an ISUP/BICC IDR		
	is present and the MCID request indicator is set to 'MCID requested'?		
	Check: Is an INFO request sent to network B and an 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?		
1	Repeat this test in reverse direction.		

Test case number	<u> </u>	pwi 001		
		SS_mwi_001		
Test case group		SIP-SIP/Service/MWI		
Reference		4.7.2/ [16]		
SELECTION EXPRESSIO		[Network A] SE 39 AND [Network B] SE 39		
Test purpose		Initial subscription of a Voicemail box.		
		ne Voicemail owner is in network A, his Voicemail box is located in network B.		
		nsure that a Voicemail owner is able to activate his Voicemail box.		
Configuration	Voice	Voicemail in network B		
	Voice	email owner in network A		
SIP Parameter	SUB	CRIBE		
		Event: message-summary		
		Expires: [any value]		
		Accept: application/simple-me	ssage-summ	nary
			0	5
	NOTI	NOTIFY		
		Subscription-State: active;expires=[any value]		
		Event: message-summary		
Message flow				
SIP (Network A)		Interconnection Interface		SIP (Network B)
		SUBCRIBE	→	
	←	200 OK SUBSCRIBE		
	_			
	←	NOTIFY		
		200 OK NOTIFY	→	
	←	200 OK BYE	_	
	_			
+		NOTIFY		
	-	200 OK NOTIFY	→	
		Apply post test routine	-	
Comments	Check:		ork A to sub	scribe to a Voicemail box in
Comments	Oneck.	Check: Is it possible for a user in network A to subscribe to a Voicemail box in network B?		
	Check:	Is the Event header in the SUE		o 'message-summary'?
	Check:			
	CHECK.			
	Chack	message-summary'?		
		Check: Is the Event header in the NOTIFY is set to 'message-summary'? Repeat this test in reverse direction.		
	Repeat t			

7.1.5.13 Message Waiting Indication (MWI)

Test case number	SS m	wi_002			
Test case group		SIP-SIP/Service/MWI			
Reference	4.7.2/	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			
				-	
				KB. Ensure that the user A is	
		nail account.	idication that there is	a new message present in his	
Configuration		mail in network B			
Comgulation		mail owner in network A			
SIP Parameter	NOTIF				
			active;expires=[any \	/alue]	
		Event: message-si			
			lication/simple-messa	ge-summary	
		Messages-Waiting: yes			
		Message-Account: sip:userA@networkA (optional)			
		Voice-Message: [any new value]/[any old value] (optional)			
Message flow					
SIP (Network A)		Interconnection		SIP (Network B)	
		200 OK INVITE	→		
		ACK	→		
		ACK	7		
		BYE	→		
		200 OK BYE	-		
	←	NOTIFY			
	:	200 OK NOTIFY	→		
		Apply post test r			
Comments	Check:	Is the Event heade	r in the NOTIFY set to	'message-summary'?	
	Check:			Y set to 'application/simple-	
	Chaole	message-summary			
	Check:				
	Check: Check:				
		Check: Contains the MIME body the optional header 'Voice-Message'? Repeat this test in reverse direction.			
	Repeat th	is test in reverse direct	011.		

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

Test case number	SS cc 002		
	SIP-SIP/Service/CC		
Test case group			
Reference	4.5.4.3/ [14]		
SELECTION EXPRESSION	[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:		
Magazza flavr	Call-Info: <sip:ue-b>;purpose=call-completion;m=NR</sip:ue-b>		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE >		
	← 180 Ringing		
	Apply post test routine		
Comments	Check: The 180 provisional response contains the Call-Info header.		
	Check: The Call-Info header contains the URI of user B as the monitor point in		
	network B.		
	eck: 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 COND is precised in a set to the astronychild and the user A involves COND.
	CCNR is possible is sent to the network A. when user A invokes CCNR,
	a SUBSCRIBE request is sent to the network B, the Event header is set to 'call-completion' and the m parameter in the Request line is set to
	'NR'.
	Ensure that the network B sends a NOTIFY request to network A to confirm that
	the request is in the Call completion queue at the terminating Application Server.
Configuration	
SIP Parameter	SUBSRIBE sip:B-AS;m= BS or m= NR
	From: <ue-a></ue-a>
	To: <ue-b></ue-b>
	Contact: <a-as></a-as>
	Event:call-completion
	NOTIFY sip:A-AS
	Event:call-completion Content-Type: application/call-completion
	state: queued
	state. queded
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	on whether CCBS or CCNR is possible is sent by network B
	SUBSCRIBE
	← 202 Accepted
	200 OK NOTIFY →
Commonto	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.

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

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

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 user B) the network B sends a NOTIFY request to network A with an indication that the subscription is terminated, the reason header is set to 'rejected'.
Configuration	
SIP Parameter	NOTIFY sip:O-AS Event:call-completion Content-Type: application/call-completion state: ready
	NOTIFY sip:O-AS
	Event:call-completion
	Subscription-State: terminated; reason=rejected
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) A CCBS or CCNR request was already successful User B is available for recall
	← NOTIFY
	200 OK NOTIFY
	CC-T9 expires
	► NOTIFY
	200 OK NOTIFY ->
Comments	Check: Is a NOTIFY request is sent to network A and the Event header is set to 'call-completion' and the state header in the message body is set to 'ready'?
	User A does not perform the recall.
	Check: Is the CC revocation is performed after timer CC-T9 expires? Repeat this test in reverse direction.

Test case number	SS_cc_007
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.2/ [14]
SELECTION EXPRESSION	([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR [Network B] SE 41)
Test purpose	User A is unavailable while CC recall is performed.
	 User A is located in network A and user B is located in network B. User A has successfully invoked a CCBS or CCNR request. User B is available for CC recal 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-recal procedure.
Configuration	
SIP Parameter	NOTIFY sip:O-AS Event:call-completion Content-Type: application/call-completion state: ready
	PUBLISH sip B-AS To: SIP 2 Event: presence Content-Type: application/pidf+xml xml version="1.0" encoding="UTF-8"? <presence <status> <basic>closed</basic></status></presence
	PUBLISH sip B-AS To: SIP 2 Event: presence Content-Type: application/pidf+xml xml version="1.0" encoding="UTF-8"? <presence <status> <basic>open</basic></status></presence
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) A CCBS or CCNR request was already successful User B is available for recall
	← NOTIFY
	200 OK NOTIFY → User A is busy PUBLISH → 200 OK PUBLISH
	User A is no longer busy PUBLISH →
	200 OK PUBLISH User B is available for recall
	← NOTIFY 200 OK NOTIFY → Apply post test routine
Comments	Check: Is a PUBLISH request is sent from Network A to Network B containing a "presence" XML element and the "basic" element is set to "closed" Check: After the User A is available again a PUBLISH request is sent
	from Network A to Network B containing a "presence" XML element and the "basic" element is set to "open"
	Repeat this test in reverse direction.

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 L 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 sen Busy Here final response and an encapsulated ISUP REL is present, th value indicator is set to #17 or #34 and the CCBS possible indicator is se 'CCBS possible'. Configuration SIP Parameter 486: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value #17 or #34 Diagnostics CCBS possible Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here (REL)	
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 u 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 sen Busy Here final response and an encapsulated ISUP REL is present, th value indicator is set to #17 or #34 and the CCBS possible indicator is set '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 CCBS possible Message flow Interconnection Interface SIP (Networf A) Interconnection Interface SIP (Networf REL)	
Test purpose SIP-I support: Indicating of CCBS possible. BICC/ISUP - SIP-I interworking applies in the terminating network and U 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 sen Busy Here final response and an encapsulated ISUP REL is present, th value indicator is set to #17 or #34 and the CCBS possible indicator is set 'CCBS possible'. Configuration	
Test purpose SIP-I support: Indicating of CCBS possible. BICC/ISUP - SIP-I interworking applies in the terminating network and U 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 sen Busy Here final response and an encapsulated ISUP REL is present, th value indicator is set to #17 or #34 and the CCBS possible indicator is set 'CCBS possible'. Configuration	
Iocated 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 sen Busy Here final response and an encapsulated ISUP REL is present, th value indicator is set to #17 or #34 and the CCBS possible indicator is se 'CCBS possible'. Configuration SIP Parameter 486: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value #17 or #34 Diagnostics CCBS possible Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here (REL)	
SIP Parameter 486: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value #17 or #34 Diagnostics CCBS possible Message flow SIP (Network A) Interconnection Interface INVITE 486 Busy Here (REL)	sends a 486 nt, the Cause
Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value #17 or #34 Diagnostics CCBS possible Message flow SIP (Network A) Interconnection Interface INVITE → € 486 Busy Here (REL)	
#17 or #34 Diagnostics CCBS possible Message flow SIP (Network A) Interconnection Interface INVITE + 486 Busy Here (REL)	
Diagnostics CCBS possible Message flow SIP (Network A) Interconnection Interface INVITE + 486 Busy Here (REL)	
Message flow SIP (Network A) Interconnection Interface SIP (Network INVITE → € 486 Busy Here (REL)	
INVITE → ← 486 Busy Here (REL)	work B)
ACK ->	
Comments Check: The 486 final response contains an encapsulated BICC/ISUP 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)
	← 180 Ringing (ACM)
	Apply post test routine
Comments	Check: The 180 provisional response contains an encapsulated ACM. Check: The CCNR possible indicator in the ACM is set to 'CCNR possible'.
	Repeat this test in reverse direction.

7.1.6 Other PSTN services (SIP-I interworking)

7.1.6.1 User-to-User Signalling (UUS)

Test case number	SS_uus_001
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] SE 47 AND SE 63
Test purpose	SIP-I support: Indicating of User-to-User service 1 implicit in initial INVITE request.
	BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 1 implicit request
	calls user B and a 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) →
2	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
	SIP-SIP/SIP-I/UUS
Test case group Reference	
SELECTION EXPRESSION	7.1, 6.5/ [24]
SELECTION EXPRESSION	([Network A] SE 17 AND [Network A] SE 47) AND ([Network B] SE 17 AND [Network B] SE 47) AND SE 63
Test purpose	SIP-I support: Indicating of User-to-User service 1 implicit response in 180 or 200 OK.
	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 a 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
	180/200
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required ACM/ANM
	User-to-user Information
	User Information
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE (IAM) →
CASE A	
	← 180 Ringing (ACM)
CASE B	← 180 Ringing (ACM)
	← 180 OK (ANM)
	Apply post test routine
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request?
	Check: Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM?
	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 or ANM?
	Repeat this test in reverse direction.

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

Test case number	SS_uus_005
	SIP-SIP/SIP-I/UUS
Test case group	
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 a User-to-user Indicator parameter is present set to 'Response', 'service 1 not provided' in the encapsulated ACM of the 180 response.
Configuration	User A is subscribed to the User-to-User service 1 explicit request User B is not subscribed to the User-to-User service 1
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM
	User-to-user Indicator Request service 1 not essential
	180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM User-to-user Indicator Response service 1 not provided
Message flow	
SIP (Network A)	 Interconnection Interface SIP (Network B) INVITE (IAM) → 180 Ringing (ACM) Apply post test routine
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request? Check: Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM? Check: Is an ISUP/BICC ACM encapsulated in the 180 response?
	Check: Is a 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 [Network A] SE 47) AND ([Network B] SE 17 AND
SELECTION EXPICESSION	[Network A] SE 47) AND SE 63
Test purpose	SIP-I support: Indicating of User-to-User service 1 essential explicit
rest purpose	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	#23 01 #03
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE (IAM) →
	← 500 Server Internal Error (REL)
	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 47AND 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 a User-to-user Indicator parameter is present set to 'Request service 2', 'not parameter's present set to 'Request service 2', 'not	
Configuration	essential' or 'essential' in the encapsulated IAM of the initial INVTE request.	
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 containing a User-to-user Indicator parameter set to Request service 2 is set to the value 'not essential' or 'essential'? Repeat this test in reverse direction.	

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

Test case group SIP-SIP/SIP-I/UUS Reference 7.1, 6.5/ [24] SELECTION EXPRESSION ([Network A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND Network B] 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 a 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	Test case number	SS_uus_009
Reference 7.1, 6.5/ [24] SELECTION EXPRESSION ([Network A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND Network B] SE 17 AND Network B] SE 47) AND SE 63 rest purpose SIP-1 support: Indicating of User-to-User service 2 not essential rejected in 180 response. BICC/ISUP - SIP-1 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 a 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 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-Type: application/isup;version=itu-t92 <th></th> <th></th>		
SELECTION EXPRESSION ([Network A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND Network B] 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 a 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 A is subscribed to the User-to-User service 2 User A is subscribed to the User-to-User service 2 Content-Type: application/isup;version=itu-192 Content-Type: application/isup;version=itu-192 Multiple ACM User-to-user Indicator		
Network BJ SE 47) AND SE 63 iest 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 a 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 User B is not subscribed to the User-to-User service 2 INVITE: Content-Type: application/isup;version=itu-192 Content-Disposition: signal;handling=required IAM User-to-user Indicator Request service 2 not essential 180 Content-Type: application/isup;version=itu-t92 Content-Type: application Iser-to-user Indicator Request service 2 not essential 180 Content-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM User-to-user Indicator Response service 2 not		
SIP-I support: Indicating of User-to-User service 2 not essential rejected in 180 response. BICC//SUP - 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 a 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 User B is not subscribed to the User-to-User service 2 INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM User-to-user Indicator	SELECTION EXPRESSION	
180 response. BICC/ISUP - SIP-1 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 a 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 INVITE: Content-Type: application/isup;version=itu-192 Content-Disposition: signal;handling=required IAM User-to-user Indicator Request service 2 not essential 180 Content-Type: application/isup;version=itu-192 Content-Disposition: signal;handling=required ACM User-to-user Indicator Request service 2 not essential 180 Content-Type: application/isup;version=itu-192 Content-Disposition: signal;handling=required ACM User-to-user Indicator Response service 2 not provided	Tost nurnoso	
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 a 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 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	l'est pulpose	
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 a 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 User B is not subscribed to the User-to-User service 2 User 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		lou response.
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 a 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 User B is not subscribed to the User-to-User service 2 User 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		BICC/ISLIP - SIP-Linterworking applies in the originating and terminating network
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 a 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 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		
user B not subscribed to User-to-User service 2 the call is rejected by the network a 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 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-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 not provided		
network a 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		
'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-Type: application/isup;version=itu-t92 Content-Type: application/isup;version=itu-t92 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 not provided		
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-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 not provided		
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 180 User-to-user Indicator Response service 2 not provided	Configuration	
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	3 1 1	
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 Parameter	
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		Content-Type: application/isup;version=itu-t92
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		
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		IAM
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		User-to-user Indicator
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		
180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM User-to-user Indicator Response service 2 not provided		service 2
Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM User-to-user Indicator Response service 2 not provided		not essential
Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM User-to-user Indicator Response service 2 not provided		
Content-Disposition: signal;handling=required ACM User-to-user Indicator Response service 2 not provided		
ACM User-to-user Indicator Response service 2 not provided		
User-to-user Indicator Response service 2 not provided		
Response service 2 not provided		
service 2 not provided		
lessage flow		
	Massage flow	Service 2 hot provided
		Interconnection Interface SIP (Network B)
INVITE (IAM) →		
← 180 Ringing (ACM)		
Apply post test routine		
	Comments	
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 a User-to-user Indicator parameter present set to 'Response',		
'service 2 not provided' in the encapsulated ISUP/BICC ACM?		'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 Network A] SE 47) AND ([Network B] SE 17 AND
	Network B] 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	#23 01 #03
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE (IAM) → 500 Server Internal Error (REL) ACK → Apply post test routine
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request?
	 Check: Is a User-to-user Indicator parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 1', 'essential'? Check: Is an ISUP/BICC REL encapsulated in the 500 response?
	Check: Is the Cause value set to #29 or #69 in the encapsulated REL?

Test case number	SS uus 011		
Test case group	SIP-SIP/SIP-I/UUS		
Reference	7.1/[24]		
SELECTION EXPRESSION	[Network A] SE 17 AND Network A] 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 a		
	User-to-user Indicator parameter is present set to 'Request service 3', 'not		
	essential' or 'essential' in the encapsulated IAM of the initial INVTE request.		
Configuration	User A is subscribed to the User-to-User service 3		
SIP Parameter	INVITE:		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	IAM		
	User-to-user Indicator		
	Request		
	service 3		
	not essential or 'essential'		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
	INVITE (IAM) →		
-	Apply post test routine		
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request		
	containing a User-to-user Indicator parameter, and the indicator		
	Request service 3 is set to the value 'not essential' or 'essential'?		
	Repeat this test in reverse direction.		

Test case number	SS_uus_	012
Test case group		SIP-I/UUS
Reference		6.5, 7.1/ [24]
SELECTION EXPRESSION	([Network	(A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND B] SE 47) AND SE 63
Test purpose	SIP-I sup success	oport: Indicating of User-to-User service 3 in initial INVITE request ful.
		JP - SIP-I interworking applies in the originating network User A is n network A and user B is located in network B.
		hen user A subscribed to the User-to-User service 3 calls user B a
		user Indicator parameter is present set to 'Request service 3', 'not ' or 'essential' in the encapsulated IAM of the initial INVTE request. The
	User-to-L	Jser service is successful.
Configuration		subscribed to the User-to-User service 3 subscribed to the User-to-User service 3
SIP Parameter	INVITE:	
		ontent-Type: application/isup;version=itu-t92 ontent-Disposition: signal;handling=required
		User-to-user Indicator
		Request
		service 3 not essential or 'essential'
	200 OK	antant Tuna: application/igup:vargion_itu t02
		ontent-Type: application/isup;version=itu-t92 ontent-Disposition: signal;handling=required
		ANM
		User-to-user Indicator
		Response service 3 provided
		Service 5 provided
	INFO	
		ontent-Type: application/isup;version=itu-t92 ontent-Disposition: signal;handling=required
		USR
		User-to-user Information
Magazara flow		User Information
Message flow SIP (Network A)		Interconnection Interface SIP (Network B)
		INVITE (IAM) →
	÷	180 Ringing (ACM)
	←	200 OK INVITE (ANM) ACK →
		INFO (USR) →
	÷	200 OK INFO
	←	INFO (USR)
		200 OK INFO →
Comments	Check:	Apply post test routine Is an ISUP/BICC IAM encapsulated in the initial INVITE request
o o miniorito	oneok.	containing a User-to-user Indicator parameter, and the indicator
		Request service 3 is set to the value 'not essential' or 'essential'?
	Check:	Is an ISUP/BICC ANM encapsulated in the 200 OK INVITE and the
		User-to-user Indicator parameter is set to 'Response', 'service 3 provided'?
	Check:	Is an ISUP/BICC USR encapsulated in the INFO message sent from
		network A to network B containing a User-to-user Information
	Check	parameter?
	Check:	Is an ISUP/BICC USR encapsulated in the INFO message sent from network B to network A containing a User-to-user Information
		parameter?
	Repeat the	nis test in reverse direction.

Test case number	SS_uus_013
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1, 6.5/ [24]
SELECTION EXPRESSION	([Network A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND
	Network B] SE 47) AND SE 63
Test purpose	SIP-I support: Indicating of User-to-User service 3 not essential rejected in 200 OK response.
	BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 3 not essential calls user B not subscribed to User-to-User service 3 the call is rejected by the network a 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
Comgulation	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 a User-to-user Indicator parameter present set to 'Response', 'service 3 not provided' in the encapsulated ISUP/BICC ANM? Repeat this test in reverse direction.

Test case number	SS_uus_014
Test case group	SIP-SIP/SIP-I/UUS
Reference	6.11.2, 7.1/ [24]
SELECTION EXPRESSION	([Network A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND
	Network B] SE 47) AND SE 63
Test purpose	SIP-I support: Indicating of User-to-User service 3 essential rejection.
	BICC/ISUP - SIP-I interworking applies in the originating and terminating network User A is located in network A and user B is located in network B. Ensure when user A subscribed to the User-to-User service 3 essential calls user B not subscribed to User-to-User service 3 the call is rejected by the network. A 500 Server Internal Error is sent and an encapsulated ISUP/BICC REL is present, the Cause value is set to #29 or #69.
Configuration	User A is subscribed to the User-to-User service 3
SIP Parameter	INVITE:
	Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM
	User-to-user Indicator
	Request
	service 3
	essential
	500
	Content-Type: application/isup;version=itu-t92
	Content-Disposition: signal;handling=required
	REL
	Cause value
	#29 or #69
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B)
	INVITE (IAM) →
	← 500 Server Internal Error (REL) ACK →
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 Network A] SE 47) AND ([Network B] SE 17 AND		
	Network B] SE 47) AND SE 63		
Test purpose	SIP-I support: Indicating of User-to-User service 3 during a session is		
	established successful.		
	BICC/ISUP - SIP-I interworking applies in the originating network User A is		
	located in network A and user B is located in network B.		
	Ensure when user A subscribed to the User-to-User service 3 user A is able to		
	request the User-to-User service 3 while the session is established. The User-to-		
	User service is successful.		
Configuration	User A is subscribed to the User-to-User service 3		
U	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		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	FAA Facility indicator		
	Facility indicator user-to-user service		
	User-to-user Indicator		
	Response		
	service 3 provided		
	Service 5 provided		
	INFO		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	USR		
	User-to-user Information		
	User Information		
Message flow			
SIP (Network A)	Interconnection Interface SIP (Network B)		
. ,	A session is already established		
	INFO (FAR)		
	★ 200 OK INFO		
	← INFO (FAA)		
	200 OK INFO		
	INFO (USR) →		
	← 200 OK INFO		
	← INFO (USR)		
	200 OK INFO		
	Apply post test routine		

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

Test case number	SS_uus_016		
Test case group	SIP-SIP/SIP-I/UUS		
Reference	5.4.3.2, 6.5, 7.1/ [24]		
SELECTION EXPRESSION	([Network A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND		
	Network B] SE 47) AND SE 63		
Test purpose	SIP-I support: Indicating of User-to-User service 3 during a session is		
	established unsuccessful.		
	BICC/ISUP - SIP-I interworking applies in the originating network User A is		
	located in network A and user B is located in network B.		
	Ensure when user A subscribed to the User-to-User service 3 user A is able to		
	request the User-to-User service 3 while the session is established. The service		
	request is rejected by Network B.		
Configuration	User A is subscribed to the User-to-User service 3		
	User B is not subscribed to the User-to-User service 3		
SIP Parameter	INFO:		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required FAR		
	Facility indicator		
	user-to-user service		
	User-to-user Indicator		
	Request		
	service 3		
	not essential		
	INFO:		
	Content-Type: application/isup;version=itu-t92		
	Content-Disposition: signal;handling=required		
	FRJ		
	Facility indicator		
	user-to-user service		
	User-to-user Indicator		
	Response		
Magage flow	service 3 not provided		
Message flow SIP (Network A)	Interconnection Interface SIP (Network B)		
SIF (Network A)	A session is already established		
	INFO (FAR) →		
	← 200 OK INFO		
	← INFO (FRJ)		
	200 OK INFO →		
	Apply post test routine		
Comments	A session is already established		
	Check: Is an ISUP/BICC FAR encapsulated in the INFO request sent from		
	Network A to Network B and a User-to-user Indicator parameter is set		
	to Is the Request service 3 'not essential'?		
	Check: Is an ISUP/BICC FAA encapsulated in the INFO request sent from		
	Network B to Network A and the User-to-user Indicator parameter is		
	set to 'Response', 'service 3 not provided'?		
	Repeat this test in reverse direction.		

7.1.6.2 Subaddressing (SUB)

Test case number	SS sub 001
Test case group	SIP-SIP/SIP-I/SUB
Reference	7.1/[24]
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] SE 47 AND SE 62
Test purpose	SIP-I support: Calling party subaddress can be correctly transferred in the Access Transport parameters.
	BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure that an ISUP/BICC ATP parameter present in the encapsulated IAM of the INVITE request and contains a Calling party subaddress.
Configuration	User A is subscribed to the SUB supplementary service
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] [any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Access transport Calling party subaddress [any boundary name]
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(IAM) → Apply post test routine
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 [Network A] SE 47 AND SE 62	
Test purpose	SIP-I support. Called party subaddress can be correctly transferred in the	
	Access Transport parameters.	
	BICC/ISUP - SIP-I interworking applies in the originating network User A is	
	located in network A and user B is located in network B. Ensure that an	
	ISUP/BICC ATP parameter present in the encapsulated IAM of the INVITE	
	request and contains a Called party subaddress.	
Configuration	User A is subscribed to the SUB supplementary service	
SIP Parameter	INVITE	
	Content-Type: multipart/mixed;boundary=[any boundary name]	
	[any boundary name]	
	Content-Type: application/isup;version=itu-t92	
	Content-Disposition: signal;handling=required	
	IAM	
	Access transport	
	Called party subaddress	
	[any boundary name]	
Message flow	la (and an un a tion la (anfana	
SIP (Network A)	Interconnection Interface SIP (Network B)	
Comments	Apply post test routine Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final	
Comments	Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response?	
	Check: Is an ISUP/BICC ATP parameter present in the encapsulated ANM	
	containing a Called party subaddress?	
	Repeat this test in reverse direction.	
L		

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 [Network B] 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	Interconnection Interface CID (Network D)
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(IAM) →
	← 180 Ringing(ACM)
	← 200 OK INVITE(ANM)
	ACK →
	Apply post test routine
Comments	Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final
	response?
	Check: Is an ISUP/BICC ATP parameter present in the encapsulated ANM
	containing a Called party subaddress?
	Repeat this test in reverse direction.

7.1.6.3	Terminal Portability (TP)
---------	---------------------------

Test case number	SS_tp_001
Test case group	SIP-SIP/SIP-I/TP
Reference	5.4.3.2/ [24]
SELECTION EXPRESSION	[Network A] SE 17 AND [Network A] 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 [Network A] 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 a BYE request is sent from Network A to Network B and an ISUP REL message is encapsulated containing a Cause value set to #102.
Configuration	User A is subscribed to the Terminal Portability supplementary service
SIP Parameter	INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required SUS Suspend/resume indicator ISDN subscriber initiated
	BYE Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Location public network serving remote user Cause value 102
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) A confirmed session already exists INFO(SUS) → € 200 OK INFO
	BYE(REL) → € 200 OK BYE
Comments	 A session is already established Check: Is an ISUP SUS message encapsulated in the INFO request and the Suspend/resume indicator set to ISDN 'subscriber initiated'? Check: Is an ISUP REL message encapsulated in the BYE request and the Cause value set to #102?
	Repeat this test in reverse direction.

7.2 Number Portability

Test case number	SS_NP_	001
Test case group	SIP-SIP/	
Reference	5.3, 5.4/	
SELECTION EXPRESSION		A] SE 13
Test purpose	Request	line in the INVITE contains the number portability indication.
	the INVIT	ttempts to call user B ported to network B. Ensure that the userinfo in FE contains a destination number in the global number format, an 'rn' er containing the Number Portability Routing Number in a global number ith hex digits and optional the 'npdi' parameter.
Configuration		
SIP Parameter	sip: +	Request line <cc> <ndc> <sn>[;npdi][; rn=(Number portability routing number)] e<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 the	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	
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 callduration callsetuptime (optional) Check the duration indicated in the CDR against the duration in the call 		
	trace.		
	7. Repeat this test in reverse direction.		

244

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

Test case number	SS_acc_003
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	Comparison of Charging Data Records > 15 min.
	Accounting of a confirmed session with a duration of > 15 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.
Configuration	
SIP Parameter	
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → € 180 Ringing € 200 OK INVITE ACK →
	Communication BYE → € 200 OK BYE
Comments	 Setup a call from network A to network B. Verify is the session confirmed. Terminate the session after 15 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 callouration 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_004
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	Comparison of Charging Data Records 25 min.
	Accounting of a confirmed session with a duration of 25 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.
Configuration	¥
SIP Parameter	
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing ← 200 OK INVITE ACK →
	Communication
	BYE →
	← 200 OK BYE
Comments	 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: called party number called party number timestamp callduration callsetuptime (optional)
	 Check the duration indicated in the CDR against the duration in the call trace.
	7. Repeat this test in reverse direction.

Test case number	22	acc 0	005			
Test case group	31	-3IF/A	CCOUNTING			
Reference						
SELECTION EXPRESSION						
Test purpose	Comparison of Charging Data Records more than 30 min. Accounting of a confirmed session with a duration of > 30 min. Verify the					
	duration of the active session stored in the CDR of both networks compared with					
	the duration in the monitored message flow at the Interconnection Interface.					
Configuration						
SIP Parameter						
Message flow						
SIP (Network A)			Interconnection Interface	SIP (Network B)		
			INVITE	→		
		←	180 Ringing			
		←	200 OK INVITE			
			ACK	→		
			Communication			
			BYE	→		
		÷	200 OK BYE			
Comments	1.	Setup	o a call from network A to net	etwork B.		
	2.	Verify	is the session confirmed.			
	3.		inate the session after 35 mi			
	4.	Deter	rmine the duration of the ses	ssion from the trace of the call monitor.		
	5.	Comp	pare the following information	n elements indicated in the CDR's of both	I	
		netwo	orks:			
		• ca	alling party number			
		• ca	alled party number			
			mestamp			
			allduration			
			allsetuptime (optional)			
	6.			ne CDR against the duration in the call		
	0.	trace.				
	7.		at this test in reverse direction	on		
	11.	i tope				

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

Test case number	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				
	ACK ->				
	Communication BYF →				
	← 200 OK BYE				
Comments	1. Setup a call from network A to network B.				
Comments	2. Verify is the session confirmed.				
	3. Terminate the session committee.				
	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 007A				
Test case group	SIP-SIP/ACCOUNTING				
Reference					
SELECTION EXPRESSION					
Test purpose	Comparison of Charging Data Records less than 1 s.				
	Accounting of a confirmed session with duration <1 s. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface.				
Configuration					
SIP Parameter					
Message flow					
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE →				
	← 180 Ringing				
	← 200 OK INVITE				
	ACK →				
	Communication				
	BYE →				
-	← 200 OK BYE				
Comments	1. Set up a call from Network A to Network B.				
	2. Verify whether the session confirmed.				
	3. 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 CDRs of both 				
	Compare the following information elements indicated in the CDRs of both networks:				
	calling party number				
	called party number				
	• timestamp				
	• call duration				
	call setup time (optional).				
	6. Check the duration indicated in the CDR against the duration in the call trace.				
	7. Repeat this test in reverse direction.				

60				
SIP-SIP/ACCOUNTING				
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.				
Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing				
BYE/CANCEL → ← 200 OK BYE/CANCEL ← 487 Request Terminated ACK →				
 Setup a call from network A to network B. Verify is an early dialogue established. Terminate the early dialogue after 20 s. Determine the duration of the session from the trace of the call monitor. Compare the following information elements indicated in the CDR's of both networks: calling party number called party number timestamp callouration callsetuptime (optional) Check the duration indicated in the CDR against the duration in the call trace. Repeat this test in reverse direction. 				

7.4 Carrier Selection

Test case number	SS csel	001
Test case group	SIP-SIP/C	
Reference		
	5.7.1.10/	
SELECTION EXPRESSION		A] SE14 AND [Network B] SE15
Test purpose	User sele	cts an operator 'call-by-call'.
		d user B are located in network A. Ensure that user A is able to call
		d user A is able to select network B as a selected carrier 'call-by-call'.
Configuration		etwork A is not presubscribed
SIP Parameter		lequest line
	sip: + <c0< th=""><th>C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc></th></c0<>	C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc>
		tequest line
	sip	: + <cc> <ndc> <sn>;npdi</sn></ndc></cc>
		[;rn= <number number="" portability="" routing="">]@<hostname>;</hostname></number>
		user=phone SIP/2.0
Message flow		
SIP (Network A)		Interconnection Interface SIP (Network B)
		INVITE 1 →
	+	INVITE 2
		Apply post test routine
Comments	Check:	Is the optional 'cic' tel uri parameter present in the Request URI in the
		INVITE sent from network A to network B identifying the selected
		carrier?
	Check:	Is the 'npdi' parameter present in the Request URI of the INVITE
		request sent from network B to network A?
	Check:	Is optional the 'rn' parameter present in the Request URI of the
		INVITE request sent from network B to network A?
	NOTE 1:	
		national agreements.
	NOTE 2:	
1		regarding the end user charging in case of Carrier selection.
	_	is test in reverse direction.

253

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.	
Reference 5.7.1.10/ [2] SELECTION EXPRESSION [Network A] SE14 AND [Network B] SE15 Test purpose User is presubscribed to operator B.	
Reference 5.7.1.10/ [2] SELECTION EXPRESSION [Network A] SE14 AND [Network B] SE15 Test purpose User is presubscribed to operator B.	
Test purpose User is presubscribed to operator B.	
Test purpose User is presubscribed to operator B.	
Linea A and user D and is actually a fact of the Aline t	
User A and user B are located in network A. Ensure that user A is able to c	all
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 SIF</hostname></sn></ndc></cc>	P/2.0
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	the state of
Comments Check: Is the optional 'cic' tel uri parameter present in the Request URI	
INVITE sent from network A to network B identifying the selected carrier?	L
Caller?	
Check: Is the 'nodi' parameter present in the Request LIPL of the IN//ITE	
Check: Is the 'npdi' parameter present in the Request URI of the INVITE request sent from petwork B to petwork A2	
request sent from network B to network A?	
request sent from network B to network A? Check: Is optional the 'rn' parameter present in the Request URI of the	
request sent from network B to network A? Check: Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A?	
request sent from network B to network A? 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 regulation	
 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 regulation national agreements. 	s or
 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 regulation national agreements. 	s or

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

Test case number	SS_csel_	004	
Test case group	SIP-SIP/C	CS	
Reference	5.7.1.10/	[2]	
SELECTION EXPRESSION	[Network	A] SE14 AND [Network B] SE15	
Test purpose		resubscribed to operator B, and overrides the preselection with	
		all via operator B.	
		•	
	User A ar	nd user B are located in network A. User A is preselected to network B.	
	Ensure th	at user A is able to call user B and user A is able to select network B	
	as a selec	cted carrier 'call-by-call'. The preselected carrier is ignored.	
Configuration	User in ne	etwork A is presubscribed to network B	
SIP Parameter	INVITE: F	Request line	
	sip: + <c0< th=""><th>C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc></th></c0<>	C> <ndc> <sn>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0</hostname></sn></ndc>	
	-		
		Request line	
	sip	o: + <cc> <ndc> <sn>;npdi</sn></ndc></cc>	
		[;rn= <number number="" portability="" routing="">]@<hostname>;</hostname></number>	
		user=phone SIP/2.0	
Message flow			
SIP (Network A)		Interconnection Interface SIP (Network B)	
	_ INVITE 1 →		
	+	INVITE 2	
	1	Apply post test routine	
Comments	Check:	Is the optional 'cic' tel uri parameter present in the Request URI in the	
		INVITE sent from network A to network B identifying the selected	
		carrier?	
	Check:	Is the 'npdi' parameter present in the Request URI of the INVITE	
		request sent from network B to network A?	
	Check:	Is optional the 'rn' parameter present in the Request URI of the	
		INVITE request sent from network B to network A?	
	NOTE 1:		
		national agreements.	
	NOTE 2:	It is possible that further information is available in the Request line	
	Bonoot th	regarding the end user charging in case of Carrier selection.	
	Repeat th	is test in reverse direction.	

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

7.5 Emergency call

Test case number	SS_ecall_0	01
Test case group	SIP-SIP/En	
Reference	5.2.10, 5.7.	1.14/ [2]
SELECTION EXPRESSION		
Test purpose	Request lin	ne in the INVITE.
	line in the II containing to present: Ge Us Na	mpts to call a PSAP located in network B. Ensure that the Request NVITE contains the emergency number and a 'rn' parameter the PSAP routing number. In addition a location information may be eolocation header and corresponding PIDF-LO Element ser-to-User header ational solution to convey location information cation information available for the PSAP.
Configuration		
SIP Parameter	INVITE: Re	quest line
	sip+	<(emergency number)>[; rn =+<(PSAP routing number)]
	@hc	ostname>;user = phone SIP/2.0
Message flow		
SIP (Network A)	I.	nterconnection Interface SIP (Network B)
		INVITE ->
-		Apply post test routine
Comments	Check: Is	s the URI in the userinfo of the Request line in a global number format
Comments	Check: Is	s the URI in the userinfo of the Request line in a global number format ontaining the PSAP routing number?
Comments	Check: Is C Check: C	s the URI in the userinfo of the Request line in a global number format
Comments	Check: Is C Check: C	s the URI in the userinfo of the Request line in a global number format ontaining the PSAP routing number? Optional: Is the URI 'rn' parameter containing the PSAP Routing Jumber?
Comments	Check: Is Check: C Check: C N Check: Is	s the URI in the userinfo of the Request line in a global number format ontaining the PSAP routing number? Optional: Is the URI 'rn' parameter containing the PSAP Routing
Comments	Check: Is Check: C Check: C N Check: Is Check: Is	s the URI in the userinfo of the Request line in a global number format ontaining the PSAP routing number? Optional: Is the URI 'rn' parameter containing the PSAP Routing Jumber? s the user parameter set to 'phone'?
Comments	Check: Is Check: C Check: K Check: Is Check: Is	s the URI in the userinfo of the Request line in a global number format ontaining the PSAP routing number? Optional: Is the URI 'rn' parameter containing the PSAP Routing lumber? s the user parameter set to 'phone'? s the location information present in the initial INVITE request.
Comments	Check: Is Check: C N Check: Is Check: Is Check: Is	s the URI in the userinfo of the Request line in a global number format ontaining the PSAP routing number? Optional: Is the URI 'rn' parameter containing the PSAP Routing Jumber? s the user parameter set to 'phone'? s the location information present in the initial INVITE request. Geolocation header PIDF-LO Element XML 'geopriv' sub element Or
Comments	Check: Is Check: C N Check: Is Check: Is Check: Is	s the URI in the userinfo of the Request line in a global number format ontaining the PSAP routing number? Optional: Is the URI 'rn' parameter containing the PSAP Routing Jumber? s the user parameter set to 'phone'? s the location information present in the initial INVITE request. Geolocation header PIDF-LO Element XML 'geopriv' sub element Or Jser-to-User header
Comments	Check: Is Check: C Check: Is Check: Is Check: Is Check: Is Check: Is	s the URI in the userinfo of the Request line in a global number format ontaining the PSAP routing number? Optional: Is the URI 'rn' parameter containing the PSAP Routing Jumber? s the user parameter set to 'phone'? s the location information present in the initial INVITE request. Geolocation header PIDF-LO Element XML 'geopriv' sub element Or Jser-to-User header Or
Comments	Check: Is Check: C Check: Is Check: Is Check: Is Check: Is Check: Is Check: Is	s the URI in the userinfo of the Request line in a global number format ontaining the PSAP routing number? Optional: Is the URI 'rn' parameter containing the PSAP Routing Jumber? s the user parameter set to 'phone'? s the location information present in the initial INVITE request. Geolocation header PIDF-LO Element XML 'geopriv' sub element Or Jser-to-User header

7.6 SIP Support of Charging

Test case number	SS_sipc	001	
Test case group	SIP-SIP/ SIP_charging		
Reference	B.2.3/ [19]		
SELECTION EXPRESSION	SE 16	•	
Test purpose	User A is in case o	ful session from user A to user B via network B one single tariff. located in network A and network B is responsible for charging (CDP) f carrier selection or service. Ensure that the network B sends a tariff on with one single tariff covered in a XML MIME body in a reliable	
		al or successful final response.	
Configuration	providiori		
SIP Parameter	INVITE:		
	18x or 20 Requ Conte	orted: 100rel 00 OK ire: 100rel entType: application/vnd.etsi.sci+xml ent-Disposition: render; handling=optional	
Mossago flow	crgt cł cł	ageType hargingControlIndicators hargingTariff tariffCurrency currentTariffCurrency communicationChargeSequenceCurrency currencyFactorScale currencyFactor currencyScale tariffDuration subTariffControl tariffControlIndicators 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: Check:	Is the supported header in the initial INVITE set to '100rel' Is the Require header in the response containing the tariff information set to '100rel'?	
	Check:	Is the messageType ' crg t' present in a 1xx provisional or a 200 OK INVITE final response?	
	Check: Check:	Is the tariffCurrency element set to 'currentTariffCurrency'? Represents the currencyFactorScale in the communicationChargeSequenceCurrency element the applicable tariff?	
	Check: Check: Repeat tl	Is the tariffDuration element set to '0'? Is the optional element ' currency ' set to 'EUR' if present? his test in reverse direction.	

259

Test case number	SS_sipc	002		
		_002 SIP_charging		
Test case group Reference				
SELECTION EXPRESSION	B.2.3/ [19 SE 16	3		
		ful accessor from upon A to upon D via notwork D coverel toriffo in		
Test purpose	one sequence of the second sec	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		
	reliable p	provisional or successful final response.		
Configuration				
SIP Parameter	INVITE: Supp	orted: 100rel		
	Conte	00 OK nire: 100rel entType: application/vnd.etsi.sci+xml ent-Disposition: render; handling=optional		
	<mark>crgt</mark>	ageType		
		nargingTariff		
		tariffCurrency		
		currentTariffCurrency		
		communicationChargeSequenceCurrency		
		currencyFactorScale		
		currencyFactor		
		currencyScale		
		tariffDuration		
		subTariffControl communicationChargeSequenceCurrency		
		currencyFactorScale currencyFactor		
		currencyScale		
		tariffDuration		
		subTariffControl		
		tariffControlIndicators		
	0	riginationIdentification		
		urrency (optional)		
Message flow	00			
SIP (Network A)		Interconnection Interface SIP (Network B) INVITE →		
CASE A	←	18x(crgt)		
	+	PRACK → 200 OK PRACK		
CASE B	+	200 OK INVITE(crgt) Apply post test routine		
Comments	Check:	Is the Supported header in the initial INVITE set to '100rel'?		
	Check:	Is the Require header in the response containing the tariff information		
		set to '100rel'?		
	Check:	Is the messageType ' crgt ' present in a 1xx provisional or a 200 OK INVITE final response?		
	Check:	Is the tariffCurrency element set to 'currentTariffCurrency'?		
	Check:	Are there more than one communicationCharge		
		SequenceCurrency elements present in the currentTariffCurrency		
		element?		
	Check:	Represents the currencyFactorScale in the communicationCharge		
		SequenceCurrency elements the applicable tariffs?		
		Is the tariffDuration element in the last applicable tariff set to '0'?		
		Is the optional element ' currency ' set to 'EUR' if present?		
	Repeat t	his test in reverse direction.		

Test case number	SS_sipc	003	
Test case group		SIP_charging	
Reference	B.2.3/ [19		
SELECTION EXPRESSION	SE 16	5]	
Test purpose		ful session from user A to user B via network B with call attempt	
	charge.		
	in case o informati	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 a call attempt charge covered in a XML MIME body in a reliable	
0	provision	al or successful final response.	
Configuration			
SIP Parameter	INVITE: Supp	orted: 100rel	
	18x or 20 Regu	00 OK ire: 100rel	
	ContentType: application/vnd.etsi.sci+xml Content-Disposition: render; handling=optional		
	mess <mark>crgt</mark>	ageType	
		nargingControlIndicators nargingTariff	
		tariffCurrency	
		currentTariffCurrency	
		communicationChargeSequenceCurrency	
	currencyFactorScale		
	currencyFactor currencyScale tariffDuration		
	subTariffControl		
		tariffControlIndicators	
		callAttemptChargeCurrency	
		currencyFactor	
	currencyScale		
	originationIdentification		
	CL	urrency (optional)	
Message flow SIP (Network A)		Interconnection Interface 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 initial INVITE set to '100rel'?	
	Check:	Is the Require header in the response containing the tariff information set to '100rel'?	
	Check:	Is the messageType a ' crgt ' present in a 1xx provisional or a 200 OK INVITE final response?	
	Check:	Is the tariffCurrency element set to 'callAttemptChargeCurrency'?	
	Check:	Represents the currencyFactorScale in the	
		callAttemptChargeCurrency element the applicable tariff?	
	Check: Repeat tl	Is the optional element ' currency ' set to 'EUR' if present? his test in reverse direction.	

Test case number	SS_sipc_	004	
Test case group		SIP_charging	
Reference	B.2.3/ [19		
SELECTION EXPRESSION	SE 16	5 <u>]</u>	
Test purpose		sful session from user A to user B via network B with call setup	
	charge.		
		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 a call setup charge covered in a XML MIME body in a reliable al or successful final response.	
Configuration	provision		
SIP Parameter	INVITE:		
SIF Farameter		orted: 100rel	
	18x or 20		
		ire: 100rel	
		entType: application/vnd.etsi.sci+xml	
	Conte	ent-Disposition: render; handling=optional	
	mess <mark>crgt</mark>	sageType	
	cł	nargingControlIndicators	
	cł	nargingTariff	
		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)	
CASE A	←	INVITE →	
CASE A	~		
	←	PRACK → 200 OK PRACK	
CASE B	←	200 OK INVITE(crgt)	
		Apply post test routine	
Comments	Check:	Is the supported header in the initial INVITE set to '100rel'?	
	Check:	Is the Require header in the response containing the tariff information set to '100rel'?	
	Check:	Is the messageType 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 t	his test in reverse direction.	

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

Test case number	SS_sipc_006
Test case group	SIP-SIP/ SIP_charging
Reference	B.2.3/ [19]
SELECTION EXPRESSION	SE 16
Test purpose	Successful change of a current tariff and next tariff during an active
rest purpose	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 <mark>crgt</mark>
	chargingControlIndicators
	chargingTariff
	tariffCurrency
	currentTariffCurrency
	communicationChargeSequenceCurrency
	currencyFactorScale
	currencyFactor
	currencyScale
	tariffDuration
	subTariffControl
	communicationChargeSequenceCurrency
	currencyFactorScale
	currencyFactor
	currencyScale
	tariffDuration
	subTariffControl
	tariffControlIndicators
	tariffSwitchCurrency
	nextTariffCurrency
	communicationChargeSequenceCurrency
	currencyFactorScale
	currencyFactor
	currencyScale
	tariffDuration
	subTariffControl
	communicationChargeSequenceCurrency
	currencyFactorScale
	currencyFactor
	currencyScale tariffDuration
	subTariffControl
	tariffControlIndicators
	tariffSwitchOverTime
	originationIdentification
	currency (optional)
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) A confirmed session already exists
	← INFO
	200 OK INFO ->
	Apply post test routine
Comments	Check: Is the messageType 'crgt' present in the INFO request?
	Check: Is the tariffCurrency element set to 'currentTariffCurrency'?
	Check: Represents the currencyFactorScale in the
	communicationChargeSequenceCurrency elements the current tariffs?
	Check: Is the tariffSwitchCurrency element set to 'nextTariffCurrency'?
	Check: Represents the currencyFactorScale in the
	communicationChargeSequenceCurrency elements the next tariffs?
	Repeat this test in reverse direction.

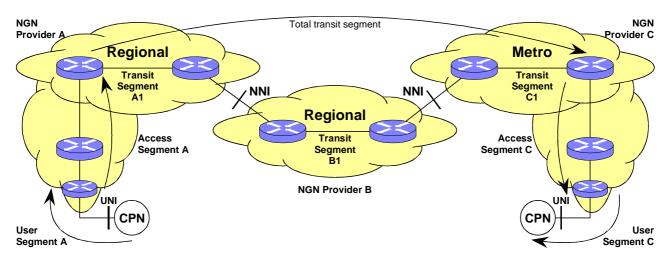
Test case number	SS_sipc_007			
Test case group	SIP-SIP/SIP_charging			
Reference	B.2.3/ [19]			
SELECTION EXPRESSION	SE 16			
Test purpose	Successful additional charge during an active session.			
	User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection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

7.7 Quality of Service (optional)

7.7.1 Delay Values

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

7.7.2 Test purposes for Quality of Service test





Test case number	SS_qos_001				
	SIP-SIP/QoS				
Test case group	0IF-0IF/QU0				
Reference					
SELECTION EXPRESSION					
Test purpose	Ensure that the UE can successfully activate the voice call via dedicated				
	bearer.				
	After establishing a voice call from the user segment A (calling user) to user				
	segment C (called user), determine the round trip delay. The called user is				
	activating a looback.				
	Based on the measurement determine the transit segment delay.				
	The call is released from the calling user.				
Configuration	The amplitude of the tone is -16 dBm0;				
configuration	Minimum uplink/downlink bandwidth is 1 Mbit/s				
SIP Parameter					
•					
Message flow					
SIP (Network A)	Interconnection Interface SIP (Network B)				
	INVITE ->				
	← 100 Trying				
	← 180 Ringing				
	← 200 OK INVITE				
	АСК →				
	Communication				
	→ BYE				
	200 OK BYE				
Comments	UE1 (a) establishes call to UE2 (b).				
	 Call answered and held for 80 seconds. 				
	Quality assessed				

Test case number	SS_qos_002			
Test case group	SIP-SIP/QoS			
Reference	[4]			
SELECTION EXPRESSION				
Test purpose	 Ensure that the UE can successfully activate the UDI data call via dedicated data bearer. User. The called user is activating a looback, the calling user is starting BER Test Based on the measurement determine the transit segment delay. The transmission quality across the exchange is unacceptable when the bit error ratio is above the alarm condition of P ≤ 10 - 5. NOTE: In Recommendation ITU-T G.826 [i.5], budgets of 18,5 % of 1,5 × 10-6 were allocated to each national network, so the packet loss for a national connection should be no more than 2,75 × 10 - 7. 			
Configuration	The call is released from the calling user.			
Configuration SIP Parameter				
Message flow SIP (Network A)	Interconnection Interface INVITE → ← 100 Trying ← 180 Ringing ← 200 OK INVITE ACK → Communication → BYE 200 OK BYE ←	SIP (Network B)		
Comments	 UE1 (a) establishes call to UE2 Call answered and held for 80 s Quality assessed 			

Test case number	SS_qos_003		
Test case group	SIP-SIP/QoS		
Reference	[4]		
SELECTION EXPRESSION			
Test purpose	Ensure that the UE can successfully activate the voice call via dedicated voice bearer. The test call is successful in the case if the Call setup time (PDD) does not exceed the values listed in table 7.2.2-1 and call is stable in unanswered and answered phases, the call remains in intelligible/high quality conversation phase for 80 seconds. 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	 UE1 (a) establishes call to UE2 (b). Call answered and held for 80 seconds. Quality assessed Repeat this test in reverse direction. 		

Table 7.7.2-1: Call setup time (post dialling delay, PDD (ETSI ES 202 765-2 (V1.2.1) [30])

Meaning of timers	Parameter	IMS, PES equivalent	IMS, PES equivalent Reference Load A		Reference Load B	
	Recommendation ITU-T Q.543 [31]					
	Detailed description		Mean Value	95 % probability of not exceeding	Mean Value	95 % probability of not exceeding
		VoLTE -VoLTE [29]				
		IMS to VoLTE				
Call setup time: The def	inition of Call setup time for VoLTE is define	d in ETSI TS 102 250-2 [32].		1	1	1
			≤ 1 950 ms	≤ 2 100 ms	≤ 2 250 ms	≤ 2 400 ms Note 1 Note 2 Note 3
	·	VoLTE to IMS (note 4)		•		
	etup time in a VoIP implementation, the time de is measured, or the time in seconds from t					
		IMS - IMS	1			
Call setup time (post dia	alling delay, PDD)					
			≤ 350 ms	≤ 500 ms	≤ 650 ms	≤ 800 ms
NOTE 3: The maximur NOTE 4: The values a	1,28 s. lane delay: 2 ms - 15 ms (S1 is the interface n value should not exceed 5,9 seconds [29]. re based on the condition that the originating tion is about 100 ms higher.			n the VoLTE - L	IE is in their state	e ECM Idle,
		FFS				

Test case number	SS_qos_004		
Test case group	SIP-SIP/QoS		
Reference	[4]		
SELECTION EXPRESSION			
Test purpose	Ensure that the UE can successfully activate the voice call via dedicated voice		
	bearer.		
	The test call is successful if the call remains in intelligible/high quality conversation phase for 80 seconds.		
	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	 UE1 (a) establishes call to UE2 (b). Call answered and held for 80 seconds. Quality assessed Repeat this test in reverse direction. 		

Annex A (informative): Test case list and test case selection

A.0 Introduction

The Excel List is an informative part of the present document. The interconnection test cases should be selected depending on the Test Suite Structure (TSS) in clause 4 and test case selection in clause 6. To provide an automatic test case selection, an Excel tool is provided and contained in archive ts_101585v020101p0.zip which accompanies the present document.

The interconnection test scenarios selection procedure is divided in four steps in this Excel file.

A.1 First step - "Identification of the Networks"

During the first step the table "Identification of the Networks" should be completed (optionally).

	Network A	Network B
Network under Test identification	Telekom Austria	Deutsche Telekom
Responsibility		
Name:	Martin Brand	Gerhard Ott
Telephone number:		
Facsimile number:		
Additional information:		
Product Supplier	Nokia	Huawei
Date of the statement:		
Dates of Testing (from to)		

Table A.1: Identification of the Networks, with examples

A.2 Second step - Selection Expression

During the second step the Selection Expression form sheet should be completed.

The Selection Expression, see clause 6 of the present document depicted in table A.2 was developed to select the scope of the compatibly test cases between network operator A and network operator B. By doing that, test purposes are selected automatically. The table may be filled out (yes/no). This table can be used like a PICS form as used in a conformance test.

The purpose of this SELECTION EXPRESSION proforma is to provide a mechanism whereby both interconnected networks may provide information about the implementation in a standardized manner		
SELECTION EXPRESSION:	Support	Support
	Network A	Network B
Network names	Telekom Austria	Deutsche Telekom
Network capabilities		
SE 1: The originating network (Network A) sends the P-Charging-Vector header?	yes	no
SE 2: The originating network (Network A) sends a subset of parameters in the P- Charging-Vector header?	no	yes
SE 3: The P-Early-Media header is supported?	no	no
SE 4: Overlap procedure using multiple INVITE method is supported?	no	no
SE 5: Overlap sending using in-dialog method is supported?	no	no
SE 6: Network supports the PSTN XML schema?	no	no
SE 7: The resource reservation procedure is supported?	yes	yes

Table A.2: Selection expression applicable in the Test Purposes with examples

271

A.3 Third step - Access and End Devises Types

During the third step the **Access and End Devises Types** form sheet should be completed. With the specified test purposes in the present document, the compatibility between the interconnected networks and the used access and end devices Selection Expression can be assured. Each Test Purpose can be performed by using a physical end device to assure end-to-end compatibility between the two interconnected networks.

71	e of End devices i	Network B		
	Telekom Austria	Deutsche Telekom		
SIP-VoIP	reletion	x		
POTS		x		
ISDN				
GSM				
VoUMTS				
VoLTE				
PSTN	x	x		
Highlight color	Explanation		Refernce	
	The user equipment is			
	SIP soft client on a PC			
		a 4G mobile device in	TS 124 229	
		The user equipment is a 3G mobile device in		
	an UMTS network			
	The user equipment is	an integrated end		
	device in the fixed ne	-	TS 183 043	
	legacy analogue devic	e		
	The user equipment is	an integrated end		
	device in the fixed ne	twork - access via a	TS 183 036	
	legacy ISDN device			
		a 2G mobile device in		
	an GSM network. SS7	SIP interworking	ITU-T Q.761 - Q764	
	applies		TS129 163	
	The user equipment is	s located in a fixed SS7	ITU-T Q.1912.5	
	network (analogue or			

A.4 Fourth step - activation

In the fourth step in the Test list (Table A.4) should be activated the filter "Selected" in row "G" (deactivate the "no" entry). In addition to hide the title of the test case deselect also the "empty" entry. Therefore row H and I are to fill in while the test execution.

Annotations to a test case (e.g. fail) could be done in the sheet "4 - Observations" in the Excel file and a reference to this in row J in sheet "3 - Test list".

Table A.4: Test list - example

IMS interconnection tests at the Ic Interface; Test Suite Structure and Test Purposes (TSS&TP)								
Test case number	Test name	Dir		Terminating end device	Selected		Verdict	Observation 💌
BCALL	SCALL							
BCALL/successful								
SS_bcall_002_a_pstn_sip	Basic call normal call clearing from the calling user.	NA -> NB	PSTN	SIP-VoIP	yes	no		2
SS_bcall_002_a_pstn_pots	Basic call normal call clearing from the calling user.	NA -> NB	PSTN	POTS	yes	no		2
SS_bcall_002_a_pstn_pstn	Basic call normal call clearing from the calling user.	NA -> NB	PSTN	PSTN	yes	no		2
SS_bcall_002_b_sip_pstn	Basic call normal call clearing from the calling user.	NB -> NA	SIP-VoIP	PSTN	yes	yes		
SS_bcall_002_b_pots_pstn	Basic call normal call clearing from the calling user.	NB -> NA	POTS	PSTN	yes	yes		
SS_bcall_002_b_pstn_pstn	Basic call normal call clearing from the calling user.	NB -> NA	PSTN	PSTN	yes	yes		
SS_bcall_003_a_pstn_sip	Request line in the INVITE.	NA -> NB	PSTN	SIP-VoIP	yes	yes		
SS_bcall_003_a_pstn_pots	Request line in the INVITE.	NA -> NB	PSTN	POTS	yes	yes		

Annex B (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".
- 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".

275

- Recommendation ITU-T Q.951: "Stage 3 description for number identification supplementary services using DSS 1".
- Recommendation ITU-T Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".
- Recommendation ITU-T Q.850 (05/98): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".
- ETSI EG 201 299-1: "Integrated Services Digital Network (ISDN); Network Integration Testing (NIT); ISDN/PSTN end-to-end testing; Part 1: Test Suite Structure and Test Purposes (TSS&TP) specification".

History

Document history			
V1.1.1	August 2012	Publication	
V1.1.2	September 2012	Publication	
V1.2.1	April 2014	Publication	
V2.1.1	February 2018	Publication	