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SERIES Q: SWITCHING AND SIGNALLING, AND ASSOCIATED MEASUREMENTS AND TESTS

Testing specifications – Testing specifications for next generation networks

NGN/IMS interconnection tests between network operators at the IMS 'Ic' interface and NGN NNI/SIP-I

Recommendation ITU-T Q.3940

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Recommendation ITU-T Q.3940

NGN/IMS interconnection tests between network operators at the IMS 'Ic' interface and NGN NNI/SIP-I

Summary

Compatibility and interoperability are key aspects of interconnection between the various national and international network operators. Consequently, it is important to aim at assuring the compatibility of user terminal equipment among the respective networks, and the interoperability of the various network entities with regard to bearer aspect and service compatibility. To help achieve this objective, Recommendation ITU-T Q.3940 describes a series of tests that could be performed as part of the interconnection process before live traffic is present.

History

Edition	Recommendation	Approval	Study Group	Unique ID*
1.0	ITU-T Q.3940	2012-08-13	11	11.1002/1000/11717
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IMS, NGN, SIP-I.

* To access the Recommendation, type the URL http://handle.itu.int/ in the address field of your web browser, followed by the Recommendation's unique ID. For example, http://handle.itu.int/11.1002/1000/11830-en.

FOREWORD

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The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

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In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database at http://www.itu.int/ITU-T/ipr/.

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NGN/IMS interconnection tests between network operators at the IMS 'Ic' interface and NGN NNI/SIP-I

1 Scope

This Recommendation defines the tests purposes (TPs) for next generation network (NGN) IP multimedia subsystem (IMS) interconnection tests between national and international network operators, covered by ITU International Telecommunication Regulations, at the IMS interconnection (Ic) interface and NGN network-to-network interface (NNI)/SIP-I. Such tests have been developed to verify the overall compatibility of the session initiation protocol (SIP), the integrated services digital network (ISDN) and the non-ISDN (public switched telephone network (PSTN)) over the national or international NGNs, with regard to the use of end devices in the relevant networks (recommended by the network operator). The test specifications cover the procedures described in [ITU-T Q.1912.5] for Profile C (SIP-I).

The specified test purposes are the basis for bilateral tests between national or international network operators. If the test between network operators is agreed, the test purposes are performed as defined in the current Recommendation. Any modification of the requirements described in, and based on, national requirements, needs additional test purposes that are not described in the current Recommendation. Any additional test may be defined and agreed between the test staff of the network operators.

This Recommendation is technically equivalent to and compatible with [b-ETSI TS 101 585].

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T Q.543] Recommendation ITU-T Q.543 (1993), *Digital exchange performance design objectives*.

[ITU-T Q.931] Recommendation ITU-T Q.931 (1993), *ISDN user-network interface layer 3 specification for basic call control.*

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

[ITU-T Q.1912.5] Recommendation ITU-T Q.1912.5 (2004), *Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part.*

[ITU-T Q.3401] Recommendation ITU-T Q.3401 (2007), *NGN NNI signalling profile (protocol set 1).*

[ITU-T Q.3630] Recommendation ITU-T Q.3630 (2017), *Inter-IMS network to network interface – Protocol specification.*

[ITU-T T.38] Recommendation ITU-T T.38 (2010), *Procedures for real-time Group 3 facsimile communication over IP networks*.

[ITU-T V.152] Recommendation ITU-T V.152 (2004), *Procedures for supporting voice-band data over IP networks*.

[ETSI TR 102 775] ETSI TR 102 775 (2011), *Speech and multimedia Transmission Quality (STQ); Guidance on objectives for Quality related Parameters at VoIP Segment-Connection Points; a support to NGN transmission planners*.

[ETSI TS 124 229] ETSI TS 124 229 V10.7.0 (2012), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE;IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP)*.

[ETSI TS 124 604] ETSI TS 124 604 V10.4.0 (2012), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication Diversion (CDIV) using IP Multimedia (IM) Core Network (CN) subsystem*.

[ETSI TS 124 605] ETSI TS 124 605 V10.0.0 (2011), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Conference (CONF) using IP Multimedia (IM) Core Network (CN) subsystem*.

[ETSI TS 124 606] ETSI TS 124 606 V10.1.0 (2011), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Message Waiting Indication (MWI) using IP Multimedia (IM) Core Network (CN) subsystem*.

[ETSI TS 124 607] ETSI TS 124 607 V10.0.0 (2011), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM) Core Network (CN) subsystem*.

[ETSI TS 124 608] ETSI TS 124 608 V10.0.0 (2011), *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*.

[ETSI TS 124 610] ETSI TS 124 610 V10.0.0 (2011), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication HOLD (HOLD) using IP Multimedia (IM) Core Network (CN) subsystem*.

[ETSI TS 124 611] ETSI TS 124 611 V10.2.0 (2012), *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*.

[ETSI TS 124 616] ETSI TS 124 616 V10.0.0 (2011), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Malicious Communication Identification (MCID) using IP Multimedia (IM) Core Network (CN) subsystem*.

[ETSI TS 124 628] ETSI TS 124 628 V10.3.0 (2011), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem*.

[ETSI TS 124 629] ETSI TS 124 629 V10.0.0 (2011), *Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Explicit Communication Transfer (ECT) using IP Multimedia (IM) Core Network (CN) subsystem*.

[ETSI TS 124 642] ETSI TS 124 642 V10.5.0 (2012), *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*.

[ETSI TS 124 654] ETSI TS 124 654 V10.1.0 (2012), *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 version 10.1.0 Release 10).*

[ETSI TS 183 036] ETSI TS 183 036 (2009), *Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); ISDN/SIP interworking; Protocol specification*.

3 Definitions

3.1 Terms defined elsewhere

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

For BICC or ISUP specific terminology, reference shall be made to [ITU-T Q.1902.2]. For SIP and SDP specific terminology, reference shall be made to [ETSI TS 124 229] and [ITU-T Q.3401], respectively. Definitions for additional terminology used in this interworking Recommendation are as given in clause 3.2

3.2 Terms defined in this Recommendation

This Recommendation defines the following terms:

3.2.1 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.

3.2.2 basic call control (BCC): Signalling protocol associated with the DSS1 – ISDN Basic Call control procedures of Recommendation [ITU-T Q.931].

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

3.2.4 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.

3.2.5 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).

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

3.2.7 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.

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

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

ACR	Anonymous Communication Rejection
ACK	Acknowledge
BICC	Bearer Independent Call Control
CB	Communication Barring
CCBS	Completion of Communications to Busy Subscriber
CCNR	Completion of Communications by No Reply
CD	Communication Deflection
CDIV	Communication Diversion
CDP	Charging Determinating Point
CDR	Communication Data Record
CFB	Communication Forwarding Busy
CFNL	Communication Forwarding Not Logged in
CFNR	Communication Forwarding No Reply
CFU	Communication Forwarding Unconditional
CONF	Conference
CUG	Closed User Group
CW	Communication Waiting
DSS1	Digital Subscriber Signalling System No. 1
ECT	Explicit Communication Transfer
GSM	Global System for Mobile Communications
GW	GateWay
HOLD	Communication Hold
Ic	Interconnection
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
IUT	Implementation Under Test

LTE	Long Term Evolution
MCID	Malicious Communication Identification
MG	Media Gateway
MWI	Message Waiting Indication
NNI	Network-to-Network Interface
OIP	Originating Identification Presentation
OIR	Originating Identification presentation Restriction
PASP	Public Answering Safety Point
PICS	Protocol Implementation Conformance Statement
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
QoS	Quality of Service
SS7	Signalling System No. 7
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIP-I	SIP with encapsulated ISUP
TIP	Terminating Identification Presentation
TIR	Terminating Identification Restriction
TP	Test Purpose
TSS	Test Suite Structure
UNI	User-to-Network Interface
UE	User Equipment
URI	Universal Resource Identifier
VoLTE	Voice over LTE

5 Conventions

This Recommendation does not use specific conventions.

6 Declarations

6.1 Reference configuration

The reference configuration depicted in Figure 6-1 shall be used to perform an interconnection test between two network operators. The reference point is depicted to observe the message flow at the IMS Ic or NGN network-to-network interface (NNI) between these two networks (called 'Interconnection Interface' in the test purposes); one for a single operator and the possible set of end devices used to perform the test purposes.

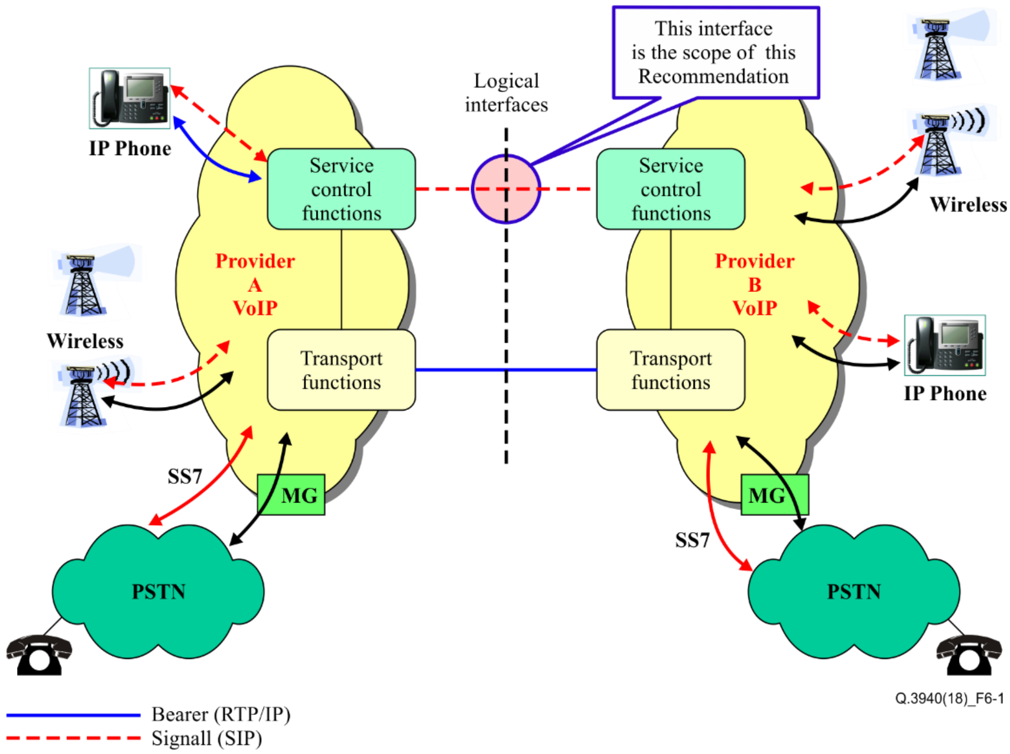

Figure 6-1 – Reference configuration for the interconnection test

6.2 Selection of end devices

Performance of the test purposes specified in this Recommendation shall assure the compatibility between the interconnected networks and the end devices that are used in the relevant networks. Each test purpose shall be performed by using a physical end device to assure the end-to-end compatibility between the two interconnected networks. This is strictly recommended due to the fact that the impact of one end device on another end device is important and will only be marginally compensated by the network.

The types of end devices that are used in the relevant network will determine which test purposes can be performed. Table 6.2-1 gives an overview of the end devices used in the relevant networks. The test staff of the network operator decides which type of end device is applicable for the test phase.

Those cells within Table 6.2-1 that contain **M** represent the mandatory type of end devices used in the test.

Those cells within Table 6.2-1 that contain **O** represent the optional type of end devices used in the test.

Table 6.2-1 – End devices used in the relevant network

Type of end devices	Network B					
Network A	**SIP**	**POTS**	**ISDN**	**GSM**	**VoLTE**	**PSTN**
SIP	**M**	**O**	**O**	**O**	**O**	**O**
POTS	**O**	**O**	**O**	**O**	**O**	**O**
ISDN	**O**	**O**	**O**	**O**	**O**	**O**
GSM	**O**	**O**	**O**	**O**	**O**	**O**
UMTS G3	**O**	**O**	**O**	**O**	**O**	**O**
VoLTE	**O**	**O**	**O**	**O**	**O**	**O**
PSTN	**O**	**O**	**O**	**O**	**O**	**O**

6.3 Selection expressions

Table 6.3-1 is used to select the optional test purposes for the compatibility test between network operator A and network operator B. The decision whether a selection expression is fulfilled is basically agreed regarding the role of the network in the test.

– Network operator 1 is in the role of Network A, and network operator 2 is in the role of Network B

In the case of mention of **Repeat this test in reverse direction** in the comment line in the test purpose:

– Network operator 2 is in the role of Network A, and network operator 1 is in the role of Network B

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

Before start of the test, Table 6.3-1 shall be completed (yes/no) to reflect responses provided by the operators to the questions asked. This table can be used as a PICS form, as used in a conformance test.

Table 6.3-1 – 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 the multiple INVITE method is supported?		
SE 5:	Overlap sending using in-dialogue method is supported?		
SE 6:	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 IWU)?		
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:	Is SIP Support of Charging supported?		
SE 17:	The interworking ISUP–SIP I is performed in the network?		

Table 6.3-1 – Selection expression applicable in the test purposes

SELECTION EXPRESSION	Support	Support
	Network A	**Network B**
SE 17a: Does the network support the session timers in the session initiation protocol (SIP)?		
SE 17b: Does the network support 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)?		
SE 28: The network supports Communication Forwarding Not Logged in (CFNL)		
SE 29: The network supports Communication Deflection?		
SE 30: The network supports the Communication Diversion (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 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)?		

Table 6.3-1 – Selection expression applicable in the test purposes

SELECTION EXPRESSION	Support	Support
	Network A	**Network B**
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 resource reservation?		
SE 43: The end device supports fax transmission via ITU-T G.711 codec?		
SE 44: The end device supports fax transmission via ITU-T V.152 codec?		
SE 45: The end device supports fax transmission via m-line ITU-T 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 user entity (UE) supports the from-change tag procedure and sends a second user identity in an UPDATE request after the dialogue is confirmed?		
SE 49: The end device performs ECT using the 'Blind/assured transfer?		
SE 50: The end device performs ECT using the 'Consultative transfer'?		
SE 51: The end device supports the Resource reservation procedure?		
PSTN/PLMN supplementary services		
SE 52: CLIP/CLIR is supported in the PSTN/PLMN part of the network?		
SE 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 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?		

Table 6.3-1 – Selection expression applicable in the test purposes

SELECTION EXPRESSION	Support Network A	Support Network B
SE 55C: CDIV Mobile subscriber not reachable is supported in the PSTN/PLMN part of the network?		
SE 55D: CDIV call defletion 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?		

7 Test purposes

The application usage procedures for the ATS shall be compliant with [ITU-T Q.3630] and [ITU-T Q.3401] respectively.

The validation of the registration procedure is out of the scope of this Recommendation.

The preconditions mechanism shall be supported by the user entity (UE) if supporting IMS.

7.1 Testing of SIP protocol requirements

7.1.1 Test purposes for basic call, successful

Test case number	SS_bcall_001
Test case group	BCALL/successful
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Basic call, normal call clearing from the called user. Ensure that call establishment is performed correctly. In the active call state, ensure the property of speech. The call is released from the called user.
Configuration	
SIP Parameter	

Message flow	
SIP (Network A) / Interconnection Interface / SIP (Network B) INVITE → ← 100 Trying ← 180 Ringing ← 200 OK INVITE ACK → Communication ← BYE 200 OK BYE →	
Comments	Establish a communication from Network A to Network B Check: Ensure the property of speech. Check: Are the media streams terminated after the 200 OK BYE was sent? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_002
Test case group	BCALL/successful
Reference	[ETSI TS 124 229]
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/[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Request line in the INVITE. Ensure that the Request line in the INVITE contains in the userpart the telephone number of the destination user equipment, formatted as a 'tel' URI in the global number format, and that 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 userpart is in the format of a tel URI in global number format. Check: The hostportion is set to the host name of the interconnected network. Check: The user parameter is set to phone. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_004
Test case group	BCALL/successful
Reference	5.10/[ETSI TS 124 229]
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 and establishes a communication between a user of Network A and a user of Network B, and that the 'icid-value' and the 'orig-ioi' parameter are present.
Configuration	
SIP Parameter	INVITE P-Charging-Vector: icid-value; orig-ioi
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ Apply post test routine	

Comments	Establish a communication from Network A to Network B. Check: The P-Charging-Vector header contains the icid-value parameter. Check: The P-Charging-Vector header contains the orig-ioi parameter. Repeat this test in reverse direction.

Test case number	SS_bcall_005
Test case group	BCALL/successful
Reference	5.10/[ETSI TS 124 229]
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 and establishes a communication between a user of Network A and a user of Network B, and subset of the parameters 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 (optional). Repeat this test in reverse direction.

Test case number	SS_bcall_006
Test case group	BCALL/successful
Reference	8/[ETSI TS 124 229]
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 and 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/[ETSI TS 124 229]			
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE3			
Test purpose	P-Early-Media header supported in early dialogue. Ensure that an early dialogue is established by sending a 183 Session Progress or 180 Ringing from Network B. Ensure that the P-Early-Media header is present and authorizes early media.			
Configuration				
SIP Parameter	INVITE P-Early-Media: supported SDP 180 P-Early-Media: [any value authorizes early media] SDP OR 183 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 sent from Network B to establish an early dialogue? Check: Is a bearer transmission possible in backward direction? (optional) NOTE 1 – The absence of the direction parameter of an 'a' line represents the default value 'sendrecv' NOTE 2 – The presence of the P-Early-Media header in the INVITE request indicates the support of "early media Authorization" in the originating Network.			

	NOTE 3 – 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.

Test case number	SS_bcall_008
Test case group	BCALL/successful
Reference	8/[ETSI TS 124 229]
SELECTION EXPRESSION	[Network A] SE 3 AND [Network B] SE 3 AND (SE 25 OR SE 26 OR SE 27 OR SE 28 OR 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 and authorizes early media. The Call is forwarded in Network B.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes
SIP Parameter	INVITE P-Early-Media: supported SDP 181 P-Early-Media: [any 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 1 – The presence of the P-Early-Media header in the INVITE request indicates the support of "early media Authorization" in the originating Network. NOTE 2 – 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/[ETSI TS 124 229]
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 and 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 1 – The presence of the P-Early-Media header in the INVITE request indicates the support of "early media Authorization" in the originating Network. NOTE 2 – 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/[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Record-route header in the INVITE. Ensure that the Via header present in the INVITE establishes a communication between a user of Network A and a user of Network B, and that the topmost header is set to the IBCF of Network A.
Configuration	
SIP Parameter	INVITE Record-Route: <Address of IBCF in Network A>
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ Apply post test routine	
Comments	Establish a communication from Network A to Network B. Check: 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/[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Via header in the INVITE. Ensure that the Via header present in the INVITE establishes a communication between a user of Network A and a user of Network B, and that the topmost header is set to the IBCF of Network A and contains a branch parameter.
Configuration	
SIP Parameter	INVITE Via: <Address of IBCF in Network A>; branch=[any value]
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ Apply post test routine	
Comments	Establish a communication from Network A to Network B. Check: The topmost Via header contains the Address of IBCF in Network A and a branch parameter. Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_012
Test case group	BCALL/successful
Reference	5.10/[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Record-Route header in the 180 Ringing. Ensure that the Record-Route header is present in the 180 Ringing provisional response as the first response from Network B; upon a connection establish setup from Network A.
Configuration	
SIP Parameter	180: Record-Route
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ 🡐 180 Ringing Apply post-test routine	
Comments	Establish a communication from Network A to Network B. Check: If the Record-Route header is present 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/[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Route header in the BYE of the originating user. Ensure that the Route header is present in the BYE request sent from the originating user equipment in Network A and that the topmost Route header or entry is set to the IBCF of Network B.
Configuration	
SIP Parameter	BYE: Route: <Address of IBCF in Network B>;lr, ….
Message flow SIP (Network A) Interconnection Interface SIP (Network B) A confirmed session already exists BYE → ← 200 OK BYE Apply post test routine	
Comments	Establish a communication from Network A to Network B. Check: Is the Route header present in the BYE, the topmost header or entry is set to the address of the IBCF of network B? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_014
Test case group	BCALL/successful
Reference	5.10/[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Route header in the BYE of the terminating user. Ensure that the Route header is present in the BYE request sent from the terminating user equipment in Network B, and that the topmost Route header or entry is set to the IBCF of Network A.
Configuration	
SIP Parameter	BYE: Route: <Address of IBCF in Network A>;lr, ….
Message flow SIP (Network A) Interconnection Interface SIP (Network B) A confirmed session already exists ← BYE 200 OK BYE → Apply post test routine	

Comments	Establish a communication from Network A to Network B Check: If the Route header present in the BYE, the topmost header or entry is set to the address of the IBCF of network A? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_015
Test case group	BCALL/successful
Reference	5.10/[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Route header in the ACK. Ensure that the Route header is present in the ACK from Network A when a connection establishment from Network A is completed, and that the topmost Route header or entry is set to the IBCF of Network B.
Configuration	
SIP Parameter	ACK: Route: <Address of IBCF in Network B>;lr, ….
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing ← 200 OK INVITE ACK → Apply post test routine	
Comments	Establish a communication from Network A to Network B. Check: 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	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Handling of SDP parameters in the INVITE. Ensure that call establishment, and the handling of the SDP parameters of the INVITE message defined as: TYPE_SDP, are 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 apply.
Configuration	

SIP Parameter	INVITE: Content-Type: application/sdp m=audio <Port number> RTP/AVP TYPE_SDP= PIXIT (Table 7.1.1-1) *or* m= Image <Port number> Udptl *or* Tcptl TYPE_SDP= PIXIT (Table 7.1.1-1) a=TYPE_SDP= PIXIT (Table 7.1.1-1) b=TYPE_SDP= PIXIT (Table 7.1.1-1)
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ Apply post test routine	
Comments	Establish a communication from Network A to Network B. Check: Is the preferred codec set to TYPE_SDP? Check: If present: is the a line set to TYPE_SDP? Check: If present: is the b line set to TYPE_SDP? Check: Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_bcall_017
Test case group	BCALL/successful
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	The SDP answer is sent in the 200 OK. Ensure that the call establishment is 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	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	First response 200 OK INVITE. Ensure that call establishment is made correctly if the called user answers with a 200 OK message.
Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 200 OK INVITE ACK → Apply post test routine	
Comments	Establish a communication from Network A to Network B Check: Is it possible to confirm a session without early dialogue? Repeat this test in reverse direction. Repeat this test with all chosen end devices

Table 7.1.1-1

TYPE_SDP	**m= line**			**b= line**	**a= line**
VA	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth -value>**	**rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>**
VA_01	audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A or rtpmap 0 PCMU/8000
VA_02	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000
VA_03	audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A or rtpmap 8 PCMA/8000
VA_04	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000
VA_05	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> CLEARMODE

Table 7.1.1-1

TYPE_SDP	**m= line**			**b= line**	**a= line**
VA	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth-value>**	**rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>**
VA_06	audio	RTP/AVP	Dynamic PT		rtpmap:<dynamic-PT> AMR-WB/16000/1
VA_07	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> AMR/8000/1
NOTE – <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.					

Test case number	SS_bcall_020
Test case group	BCALL/successful
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	[Network A] SE 43 AND [Network B] SE 43
Test purpose	Fax transmission using the ITU-T G.711 codec. Ensure that a fax transmission is possible from Network A to Network B and that the relevant codec is the ITU-T G.711 codec. Ensure in the active call state the property of fax transmission. The call establishment procedures based on SIP/SDP and ITU-T H.248 for a real-time fax over IP service are described in ITU-T Q.4016.
Configuration	
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8/0 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP 8
Message flow SIP (Network A)	Interconnection Interface — SIP (Network B) INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → Apply post test routine
Comments	Establish a communication from Network A to Network B. Check: Is the SDP answer contained in the 200 OK INVITE? Check: Is fax transmission successful? Repeat this test in reverse direction.

Test case number	SS_bcall_021
Test case group	BCALL/successful
Reference	[ETSI TS 124 229], [ITU-T V.152]
SELECTION EXPRESSION	[Network A] SE 44 AND [Network A] SE 44

Test purpose	Fax transmission using the ITU-T V.152 codec. Ensure that a fax transmission is possible from Network A to Network B and the relevant codec is the ITU-T V.152 codec. Ensure in the active call state the property of fax transmission. The call establishment procedures based on SIP/SDP and ITU-T H.248 for a real-time fax over IP service are described in ITU-T Q.4016.
Configuration	
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8 <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes 180/200 OK INVITE: SDP m=audio <Port> RTP/AVP <dynamic-PT> a=rtpmap <dynamic-PT> PCMA/8000 a=gpmd; vbd=yes
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → Apply post test routine	
Comments	Establish a communication from Network A to Network B. Check: Contains the SDP offer in the initial INVITE a voice band data codec. Check: Contains the SDP answer in the 180 or 200 OK INVITE a voice band data codec. Check: Is fax transmission successful? Repeat this test in reverse direction.

Test case number	SS_bcall_022
Test case group	BCALL/successful
Reference	[ETSI TS 124 229], [ITU-T T.38]
SELECTION EXPRESSION	[Network A] SE 45 AND [Network B] SE 45
Test purpose	Fax transmission using the ITU-T T.38 in an audio m-line codec. Ensure that a fax transmission is possible from Network A to Network B, and that the relevant codec is the ITU-T 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 ITU-T H.248 for a real-time fax over IP service are described in ITU-T Q.4016.
Configuration	
SIP Parameter	INVITE: SDP m=audio <Port> RTP/AVP 8 OR <dynamic-PT> a=rtpmap 8 OR <dynamic-PT> PCMA/8000 m=image <Port> udptl t38 180/200 OK INVITE: SDP m=image <Port> udptl t38

Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE (SDP1) → ← 180 Ringing ← 200 OK INVITE (SDP2) ACK → Apply post test routine	
Comments	Establish a communication from Network A to Network B. Check: Contains the SDP offer in the initial INVITE, an ITU-T T.38 codec in an 'audio' line. Check: Contains the SDP answer in the 180 or 200 OK INVITE and ITU-T T.38 codec in an 'audio' line. Check: Is fax transmission successful? Repeat this test in reverse direction.

Test case number	SS_bcall_023
Test case group	BCALL/successful
Reference	4.9, N/[ETSI TS 124 229]
SELECTION EXPRESSION	[Network A] SE 47 AND [Network A] SE 4 AND [Network B] SE 4
Test purpose	Overlap sending, the multiple INVITE method is used. Ensure that call establishment using overlap sending is performed correctly. Ensure that in the confirmed state, the voice transfer on the media and B-channels is performed correctly.
Configuration	
SIP Parameter	
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(CSq 1) → INVITE(CSq 2) → ← 484 Address Incomplete(CSq 1) ACK → INVITE(CSq 3) → ← 484 Address Incomplete(CSq 2) ACK → ……. INVITE(CSq 4) → ← 484 Address Incomplete(CSq 3) ACK → ← 180 Ringing(CSq 4) Apply post test routine	

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

<table>
<tr><td>Test case number</td><td>SS_bcall_024</td></tr>
<tr><td>Test case group</td><td>BCALL/successful</td></tr>
<tr><td>Reference</td><td>4.9, N/[ETSI TS 124 229]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 47 AND [Network A] SE 4 AND [Network B] SE 5</td></tr>
<tr><td>Test purpose</td><td>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.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td>INVITE 2:
Supported: 100rel
183: Require: 100rel
INFO:
Content-Type: application/x-session-info
SubsequentDigit: <additional digits></td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(CSq 1) 1 →
← 484 Address Incomplete(CSq 1)
ACK →

INVITE(CSq 2) 2 →
← 183 Session Progress(CSq 2)
PRACK →
← 200 OK PRACK

INFO →
← 200 OK INFO

……..

INFO →
← 200 OK INFO

← 180 Ringing(CSq 2)
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Establish a communication from ISDN to SIP using the overlap operation in ISDN.</td></tr>
</table>

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

Test case number	SS_bcall_025
Test case group	BCALL/successful
Reference	5.1.1.1.2/[ETSI TS 183 036]
SELECTION EXPRESSION	[Network A] (SE 46 OR SE 47) AND [Network A] SE 6
Test purpose	PSTN XML BearerCapability element in the INVITE. User A is located in Network A and an ISDN end device is used. Ensure that the INVITE request contains a PSTN XML MIME body and a BearerCapability element as indicated in Table 7.1.1-2 is present.
Configuration	User A is an ISDN access either in the PSTN or the SIP – ISDN Interworking according to [b-ETSI TS 124 615] applies
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>ITC_value< < BCoctet4 TransferMode>00< InformationTransferRate>10000< BCoctet5 Layer1Identification>01< UserInfoLayer1Protocol>00011<
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ Apply post test routine	
Comments	Check: Is a PSTN XML MIME body contained in the INVITE request? Check: Is the BearerCapability element is present? Check: Is InformationTransferCabability element is set as indicated in Table 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 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/[ETSI TS 183 036]
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 to [b-ETSI TS 124 615] applies
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN HighLayerCompatibility HLOctet3 CodingStandard>00< Interpretation>100< PresentationMethod>01< HLOctet4 HighLayerCharacteristics>[any value]<
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ Apply post test routine	
Comments	Check: Is a PSTN XML MIME body contained in the INVITE request? Check: Is the HighLayerCapability element is present? Repeat this test in reverse direction.

Test case number	SS_bcall_027
Test case group	BCALL/successful
Reference	5.1.1.1.2/[ETSI TS 183 036]
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 [b-ETSI TS 124 615] applies
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN ProgressIndicator ProgressOctet3 CodingStandard>00< Location>yyyy< ProgressOctet4 ProgressDescription>0000110< *ProgressIndicator* *ProgressOctet3* *CodingStandard>00<* *Location>0000<* *ProgressOctet4* *ProgressDescription>[any value]<*
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ Apply post test routine	
Comments	Check: Is a PSTN XML MIME body contained in the INVITE request? Check: Is a ProgressIndicator element present and the ProgressDescription element is set to '0000110'? Check: Is optional a second ProgressIndicator element present and the ProgressDescription element is set to any value not #2 and not #8? Repeat this test in reverse direction.

Test case number	SS_bcall_028
Test case group	BCALL/successful
Reference	5.1.2.2/[ETSI TS 183 036]
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 [b-ETSI TS 124 615] applies
SIP Parameter	180: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional

	<?xml version="1.0" encoding="utf-8"?> PSTN ProgressIndicator ProgressOctet3 CodingStandard>00< Location>yyyy< ProgressOctet4 ProgressDescription>0000111< *ProgressIndicator* *ProgressOctet3* *CodingStandard>00<* *Location>0000<* *ProgressOctet4* *ProgressDescription>[any value]<*
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing Apply post test routine	
Comments	Check: Is a PSTN XML MIME body contained in the 180 Ringing response? Check: Is a ProgressIndicator element present and is the ProgressDescription element set to '0000111'? Check: Is (optional) a second ProgressIndicator element present and is the ProgressDescription element 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/[ETSI TS 183 036]
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 that 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 [b-ETSI TS 124 615] 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 is the ProgressDescription element set to '0000111'? Repeat this test in reverse direction.

Test case number	SS_bcall_030
Test case group	BCALL/successful
Reference	5.1.1.1.2/[ETSI TS 183 036]
SELECTION EXPRESSION	[Network A] (SE 46 OR SE 47) AND [Network A] SE 6
Test purpose	PSTN XML BearerCapability Fallback connection type element in the INVITE. User A is located in Network A and an ISDN end device is used. Ensure that the INVITE request contains a PSTN XML MIME body and one BearerCapability element is present. The InformationTransferCabability element is set to '00000' and the one InformationTransferCabability element is set to '10001'.
Configuration	User A is an ISDN access either in the PSTN or the SIP – ISDN Interworking according to [b-ETSI TS 124 615] applies
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>00000< BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>10001<
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → Apply post-test routine	

Comments	Check: Is a PSTN XML MIME body contained in the INVITE request? Check: Is the first BearerCapability InformationTransferCabability element is set as indicated to '00000'? Check: Is the second BearerCapability InformationTransferCabability element is set as indicated to '10001'? Check: Is the InformationTransferCabability element value consistent with the codec list in the SDP? Check: Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP? Repeat this test in reverse direction.

Test case number	SS_bcall_031
Test case group	BCALL/successful
Reference	5.1.2.3/[ETSI TS 183 036]
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, that a BearerCapability element is present and that the InformationTransferCabability element is set to '10001'.
Configuration	User B is an ISDN access either in the PSTN or the SIP – ISDN Interworking according to [b-ETSI TS 124 615] applies
SIP Parameter	200: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>10001<
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE → ← 180 Ringing ← 200 OK INVITE ACK → Apply post test routine	
Comments	Check: Is a PSTN XML MIME body contained in the 200 OK INVITE response? Check: Is a BearerCapability element present, and 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/[ETSI TS 183 036]
SELECTION EXPRESSION	[Network B] (SE 46 OR SE 47) AND [Network B] SE 6
Test purpose	Fall back occurs. User B is located in Network B and an ISDN end device is used. The Fallback connection type was requested in the initial INVITE request. Ensure that the 200 OK INVITE response contains a PSTN XML MIME body, that a BearerCapability element is present and that the InformationTransferCabability element is set to '00000'. A PSTN XML MIME ProgressIndicator body is present, the ProgressDescription is set to '0000101'.
Configuration	User B is an ISDN access either in the PSTN or the SIP – ISDN Interworking according to [b-ETSI TS 124 615] applies
SIP Parameter	200: Content-Type: application/vnd.etsi.pstn+xml Content-Disposition: signal;handling=optional <?xml version="1.0" encoding="utf-8"?> PSTN BearerCapability BCoctet3 CodingStandard>00< InformationTransferCabability>00000< ProgressIndicator ProgressOctet4 ProgressDescription>0000101<
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing ← 200 OK INVITE ACK → Apply post test routine	
Comments	Check: Is a PSTN XML MIME body contained in the 200 OK INVITE response? OR If no PSTN XML MIME body contained in the 200 OK INVITE response, No further checks. Check: Is a BearerCapability element present, and the InformationTransferCabability element set to '00000'? Check: Is a ProgressIndicator element is present, and the ProgressDescription is set to '0000101' Check: Is the InformationTransferCabability element value consistent with the codec list in the SDP? Check: Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP? Repeat this test in reverse direction.

Test case number	SS_bcall_032A
Test case group	BCALL/successful
Reference	5.1.2.3/[b-IETF RFC 4733]
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> NOTIFY CASE B m=audio <Port> RTP/AVP <dynamic-PT> a=rtpmap <dynamic-PT> telephone-event/8000 a=rtpmap <dynamic-PT> 0-15 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 TMF 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 TMF signal contain the events and duration regarding the used content type? Repeat this test in reverse direction.

Test case number	SS_bcall_032B
Test case group	BCALL/successful
Reference	[ETSI TS 124 628]
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 handeled 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. Repeat this test in reverse direction.

Test case number	SS_bcall_033
Test case group	BCALL/successful
Reference	7.1/[ITU-TQ.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47
Test purpose	SIP-I support, Basic call, IAM present in the INVITE request. Ensure that when a call initiated in the PSTN or the PLMN and the ISUP – SIP-I interworking is applicable in the originating network, an ISUP IAM is encapsulated in the initial INVITE request. Ensure that all the mandatory parameters in the IAM are present and that the values are valid and the Transmission medium requirement parameter is consistent with the SDP.
Configuration	
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Nature of connection indicators Forward call indicators Calling party's category Transmission medium requirement Called party number *Calling party number (optional)* *Optional forward call indicators (optional)* *Hop counter (optional)* *User service information (optional)* *Access transport (optional)* --[any boundary name]--
Message flow	
SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(IAM) ➔ 🡐 100 Trying Apply post test routine
Comments	Establish a communication from Network A to Network B Check: Is an ISUP IAM encapsulated in the INVITE request? Check: Are all the mandatory ISUP parameters present in the IAM and are the values valid? Check: Are the values of the optional parameters in the encapsulated IAM valid? Check: Is the 'm' line with corresponding attributes in the SDP consistent with the Transmission medium requirement parameter?

	Check: Is the Transmission medium requirement value consistent with the bandwidth information in the SDP? Repeat this test in reverse direction.

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	BCALL/successful
Reference	7.2.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 4 AND SE 17 AND SE 47
Test purpose	SIP-I support, Basic call, overlap signalling. Ensure that when overlap signalling applies in the ISUP -SIP-I interworking in the originating network, several INVITE requests with the same Cal-ID and From tag are sent from Network A to Network B. Ensure that the original IAM is encapsulated in any INVITE request.
Configuration	
SIP Parameter	
Message flow	

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

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

Test case number	SS_bcall_035
Test case group	BCALL/successful
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	SIP-I support, Basic call, ACM present in the 180 response. Ensure that on receipt of a 180 Ringing provisional response an SIP-I – ISUP interworking is applicable in the terminating network, the Backward call indicators parameter in the encapsulated ACM is present, and the values are valid. Ensure that the values of the optional parameters in the encapsulated ACM are valid.
Configuration	
SIP Parameter	180: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicators 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? Repeat this test in reverse direction.

Test case number	SS_bcall_036
Test case group	BCALL/successful
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	SIP-I support. Basic call, early ACM present in the 183 response. Ensure that on receipt of a 183 Session Progress provisional response and an SIP-I – ISUP interworking is applicable in the terminating network, that 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 Acces 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/[Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	SIP-I support. Basic call, CPG present in a 180 response. Ensure that on receipt of a 180 Ringing provisional response, and an SIP-I – ISUP interworking is applicable in the terminating network, the Event indicator in the encapsulated CPG is present and set to 'ALERTING'. Ensure that the values of the optional parameters in the encapsulated CPG are valid.
Configuration	Select a proper destination that sends at first an early ACM and after then a CPG 'ALERTING' in the PSTN/PLMN (e.g., PBX).
SIP Parameter	180: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Event indicator = ALERTING --[any boundary name]--
Message flow	
SIP (Network A)	Interconnection Interface / SIP (Network B) INVITE → ← 100 Trying ← 183 Session Progress(ACM) ← 180 Ringing(CPG) Apply post test routine
Comments	Establish a communication from Network A to Network B Check: Is an ISUP CPG message encapsulated in the 180 Ringing provisional response? Check: Is the mandatory Event indicator present in the encapsulated ISUP CPG set to 'ALERTING'? Check: Are the values of optional parameters in the encapsulated ISUP CPG valid? Repeat this test in reverse direction.

Test case number	SS_bcall_038
Test case group	BCALL/successful
Reference	6.7/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47

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

Test case number	SS_bcall_039
Test case group	BCALL/successful
Reference	5.4.3.4, 6.11.2/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47
Test purpose	SIP-I support. Basic call, REL present in a BYE request sent from the originating network. Ensure that a ISUP REL message is encapsulated in a BYE request sent in the release procedure initiated from the originating user when ISUP – SIP-I interworking is applicable in the originating network. Ensure the validity of the cause indicator in the encapsulated REL. Ensure that the ISUP RLC is encapsulated in the 200 OK BYE.
Configuration	

<table>
<tr><td>SIP Parameter</td><td colspan="2">BYE:
Content-Type: multipart/mixed;boundary=[any boundary name]
--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
REL
Cause value:
--[any boundary name]--

200 OK BYE
Content-Type: multipart/mixed;boundary=[any boundary name]
--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
RLC
--[any boundary name]--</td></tr>
<tr><td colspan="3">Message flow</td></tr>
<tr><td>SIP (Network A)</td><td>Interconnection Interface</td><td>SIP (Network B)</td></tr>
<tr><td></td><td>INVITE →</td><td></td></tr>
<tr><td></td><td>← 100 Trying</td><td></td></tr>
<tr><td></td><td>← 180 Ringing</td><td></td></tr>
<tr><td></td><td>← 200 OK INVITE</td><td></td></tr>
<tr><td></td><td>ACK →</td><td></td></tr>
<tr><td></td><td>Communication</td><td></td></tr>
<tr><td></td><td>BYE(REL) →</td><td></td></tr>
<tr><td></td><td>← 200 OK BYE(RLC)</td><td></td></tr>
<tr><td>Comments</td><td colspan="2">Establish a confirmed communication from Network A to Network B.
The originating user terminates the communication.
Check: Is the ISUP REL encapsulated in the BYE request?
Check: Are the cause indicators in the encapsulated ISUP REL valid?
Check: If a Reason header is present in the BYE request, is the 'cause' value of Reason header equal to the 'cause value' in the encapsulated REL?
Check: Is the ISUP RLC encapsulated in the 200 OK BYE?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_bcall_040
Test case group	BCALL/successful
Reference	5.4.3.4, 6.11.2/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	SIP-I support. Basic call, REL present in a BYE request sent from the terminating network Ensure that 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 REL Cause value: --[any boundary name]-- 200 OK BYE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required RLC --[any boundary name]--
Message flow	

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

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

7.1.2 Codec negotiation

Test case number	SS_codec_001
Test case group	BCALL/Codec_Negotiation
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	

Test purpose	Session update requested by the calling user. During the session, the calling user decides to change the characteristics of the media session. This is accomplished by sending a re-INVITE or UPDATE containing a new media description. This re-INVITE or UPDATE references the existing 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. When the parameter in the SDP rtpmap:<dynamic-PT> is used, the codecs in Table 7.1.2-1 apply.
Configuration	
SIP Parameter	SDP1: codec x chosen from Table 7.1.2-1 SDP3: codec y chosen from Table 7.1.2-1
Message flow SIP (Network A) Interconnection Interface SIP (Network B) A confirmed session already exists (SDP 1) CASE A INVITE(SDP3) → ← 200 OK INVITE(SDP4) ACK → CASE B UPDATE(SDP3) → ← 200 OK UPDATE(SDP4) Apply post test routine	
Comments	Establish a communication from Network A to Network B using SDP1 chosen from the Table 7.1.2-1. Check: The calling user changes the media description using INVITE request containing SDP 3 codec chosen from Table 7.1.2-1, different to SDP1. Check: Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description? Repeat this test in reverse direction.

Test case number	SS_codec_002
Test case group	BCALL/Codec_Negotiation
Reference	[ETSI TS 124 229]
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. When the parameter in the SDP rtpmap:<dynamic-PT> is used, the codecs in Table 7.1.2-1 apply.

Configuration	
SIP Parameter	SDP1: codec x chosen from Table 7.1.2-1 SDP2: codec y chosen from Table 7.1.2-1
Message flow SIP (Network A) / Interconnection Interface / SIP (Network B) A confirmed session already exists (SDP 1) CASE A ← INVITE(SDP3) → 200 OK INVITE(SDP4) ACK → CASE B UPDATE(SDP3) → ← 200 OK UPDATE(SDP4) Apply post test routine	
Comments	Establish a connection from SIP UE 1 to SIP UE 2 using SDP1 chosen from 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	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	The SDP answer is contained in a 200 OK final response. Ensure that the call establishment is performed correctly. The initial INVITE contains an SDP with the offer 1. Ensure that the answer related to the SDP offer is contained in the 200 OK INVITE message. Ensure that in the confirmed call state the voice transfer on the media channels is performed correctly.
Configuration	
SIP Parameter	INVITE: SDP offer 200: SDP answer
Message flow SIP (Network A) / Interconnection Interface / SIP (Network B) INVITE(SDP1) → ← 180 Ringing ← 200 OK INVITE(SDP2) ACK → Apply post test routine	

Comments	Establish a communication from Network A to Network B Check: Is the SDP offer contained in the initial INVITE request? Check: Is the SDP answer contained in the 200 OK INVITE final response? Repeat this test in reverse direction.

Table 7.1.2-1

VARIABLE	PT	Encoding	media type	clock rate	channels	Supported in network A	Supported in network B
VA_01	0	PCMU	A	8,000	1		
VA_02	3	GSM	A	8,000	1		
VA_03	4	G723	A	8,000	1		
VA_04	5	DVI4	A	8,000	1		
VA_05	6	DVI4	A	16,000	1		
VA_06	7	LPC	A	8,000	1		
VA_07	8	PCMA	A	8,000	1		
VA_08	9	G722	A	8,000	1		
VA_09	10	L16	A	44,100	2		
VA_10	11	L16	A	44,100	1		
VA_13	12	QCELP	A	8,000	1		
VA_12	13	CN	A	8,000	1		
VA_13	14	MPA	A	90,000			
VA_14	15	G728	A	18,000	1		
VA_15	16	DVI4	A	11,025	1		
VA_16	17	DVI4	A	22,050	1		
VA_17	18	G729	A	8,000	1		
VA_18	Dyn	G726-40	A	8,000	1		
VA_19	Dyn	G726-32	A	8,000	1		
VA_20	Dyn	G726-24	A	8,000	1		
VA_21	Dyn	G726-16	A	8,000	1		
VA_22	Dyn	G729D	A	8,000	1		
VA_23	Dyn	G729E	A	8,000	1		
VA_24	Dyn	GSM-EFR	A	8,000	1		
VA_25	25	CelB	V	90,000			
VA_26	26	JPEG	V	90,000			
VA_27	28	Nv	V	90,000			
VA_28	31	H261	V	90,000			
VA_29	32	MPV	V	90,000			

Table 7.1.2-1

VARIABLE	PT	Encoding	media type	clock rate	channels	Supported in network A	Supported in network B
VA_30	33	MP2T	V	90,000			
VA_31	34	H263	V	90,000			
VA_32	Dyn	H263-1998	V	90,000			
VA_33	Dyn	AMR	A	8,000	1		
VA_34	Dyn	AMR-WB	A	16,000	1		
VA_35	Dyn	telephone-event	A	8000	1		

7.1.3 Resource reservation

Test case number	SS_resource_001
Test case group	BCALL/Resource_Reservation
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	([Network A] SE 50 AND [Network B] SE 50) AND SE 7
Test purpose	Resource reservation successful, segmented status. Ensure that the network is able to reserve resources for quality of service (QoS) 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	

<table>
<tr><td>SIP Parameter</td><td>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
183 Session Progress: Supported: 100rel precondition
SDP2: 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 mandatory/optional remote sendrecv
UPDATE
SDP3: m=audio <Port number> RTP/AVP <codec>
a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory/optional local sendrecv
a=des:qos mandatory/optional remote sendrecv
200 OK UPDATE
SDP4: a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory/optional local sendrecv
a=des:qos mandatory/optional remote sendrecv</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(SDP1) →
← 183 Session Progress(SDP2)
PRACK →
← 200 OK PRACK
Resource reservation
UPDATE(SDP3) →
← 200 OK UPDATE(SDP4)
Apply post test routine</td></tr>
<tr><td>Comments</td><td>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/optional ' and 'sendrecv' in the 183?
Check: Is the quality of service for the current state local set to 'sendrecv' indicated in the SDP in the UPDATE?
Check: Is the quality of service for the current state local and remote set to 'sendrecv' indicated in the SDP in the 200 OK UPDATE?
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.
NOTE – This test case is applicaple with a VoLTE originator and termination
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_resource_002
Test case group	BCALL/Resource_Reservation
Reference	[b-IETF RFC 4566], [b-IETF RFC 3261], [b-IETF RFC 3264], [b-IETF RFC 3312]
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 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>

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 applicaple with a VoLTE originator

7.1.4 Test purposes for SIP-SIP, basic call, unsuccessful

Test case number	SS_unsucc_001
Test case group	BCALL/unsuccessful
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Called number is not allocated in the assumed network. Ensure that, when calling to unallocated number, the network initiates call clearing to the calling user with a 404 Not Found message.
Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 404 Not Found ACK →	
Comments	Establish a communication from Network A to Network B, called user number is not allocated in Network B Check: Is a 404 Not Found sent from Network B to Network A? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_unsucc_002
Test case group	BCALL/unsuccessful
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	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	Establish a communication from Network A to Network B, Network B is unable to process the request. Check: Is a 503 Service unavailable sent from Network B to Network A? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_unsucc_003
Test case group	BCALL/unsuccessful
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	The called user is network determined busy. Ensure that, when the called user is busy, the network initiates call clearing to the calling user with a 486 Busy Here message.
Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 486 Busy Here ACK →	
Comments	Establish a communication from Network A to Network B, user B is network determined user busy. Check: Is a 486 Busy Here sent from Network B to Network A? Repeat this test in reverse direction.

Test case number	SS_unsucc_004
Test case group	BCALL/unsuccessful
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	The called user is user determined busy. Ensure that, when the called user is busy, the user initiates call clearing to the calling user with a 486 Busy Here message.
Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 486 Busy Here ACK →	
Comments	Establish a communication from Network A to Network B, user B is user determined user busy. Check: Is a 486 Busy Here sent from Network B to Network A? Repeat this test in reverse direction.

Test case number	SS_unsucc_005
Test case group	BCALL/unsuccessful
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	The called user is not available on the called number. Ensure that when the number is changed, the network initiates call clearing to the calling user with a 410 Gone message.
Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ 🡐 410 Gone ACK ➔	
Comments	Establish a communication from Network A to Network B, user B is not allocated in Network B. Check: Is a 410 Gone sent from Network B to Network A? Repeat this test in reverse direction.

Test case number	SS_unsucc_006
Test case group	BCALL/unsuccessful
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	The number of the called user is incomplete. Ensure that the call will be released when the called number is incomplete. The network initiates call clearing to the calling user with 484 Not Found message.
Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ 🡐 484 Address Incomplete ACK ➔	
Comments	Establish a communication from Network A to Network B, the called number is incomplete. Check: Is a 484 Address Incomplete sent from Network B to Network A? Repeat this test in reverse direction.

Test case number	SS_unsucc_007
Test case group	BCALL/unsuccessful
Reference	[ETSI TS 124 229]
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 an 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	[ETSI TS 124 229]
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 ← INVITE 488 Not Acceptable Here → ← ACK Apply post test routine
Comments	Establish a communication from Network A to Network B. User B in Network B attempts to change the session by sending an SDP offer to the UE in Network A. Network A does not support the codec sent in the offer. Check: Is a 488 Not Acceptable Here sent from Network B to Network A? Repeat this test in reverse direction.

Test case number	SS_unsucc_009
Test case group	BCALL/unsuccessful
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Call clearing due to no answer from the called user initiated by the calling user. Ensure that when there is no answer from the called user, the calling user initiates call clearing to the called user with CANCEL or BYE.
Configuration	
SIP Parameter	

Message flow SIP (Network A)	Interconnection Interface INVITE ➔ 🡐 180 Ringing CANCEL/BYE ➔ 🡐 200 OK CANCEL/BYE 🡐 487 Request Terminated ACK ➔	SIP (Network B)
Comments	Check: Is a CANCEL or BYE request sent by the the originating user? Check: Is a 487 Request Terminating sent by the the terminating user? Check: Are the media streams terminated after the 200 OK CANCEL/BYE was sent? Repeat this test in reverse direction.	

Test case number	SS_unsucc_010
Test case group	BCALL/unsuccessful
Reference	[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Codec not supported by the called user. The initial INVITE contains an SDP with codecs that are not supported by the called user. Ensure that, when the called user does not accept the Media session, the called user initiates call clearing to the calling user with 488 Not Acceptable Here or 606 Not Acceptable, which also receives an ACK.
Configuration	
SIP Parameter	INVITE: codec not supported at user (Network B)
Message flow SIP (Network A) Interconnection Interface SIP (Network B) ➔ INVITE ➔ CASE A 🡐 488 Not Acceptable Here 🡐 ➔ ACK ➔ CASE B 🡐 606 Not Acceptable 🡐 ➔ ACK ➔	
Comments	Establish a call setup from Network A to Network B. User B in Network B does not support the codec offered in the SDP received from Network A. Check: Is a 488 Not Acceptable Here or 606 Not Acceptable sent from Network B to Network A? Repeat this test in reverse direction.

Test case number	SS_unsucc_011
Test case group	BCALL/unsuccessful
Reference	[ETSI TS 124 229]
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 ➔ 🡐 200 OK CANCEL/BYE 🡐 487 Request Terminated ACK ➔
Comments	Check: Is a CANCEL or BYE request sent by the originating network? Check: Is a 487 Request Terminating sent 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	[b-IETF RFC 4028]
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 to short for one of the involved entities, a 422 Session Interval Too Small unsuccessful final response is sent in backward direction to update the session duration time. A new INVITE 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/[ITU-T Q.1912.5]
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, that the network initiates 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 encapsulated ISUP REL? Repeat this test in reverse direction.

Test case number	SS_unsucc_013
Test case group	BCALL/unsuccessful
Reference	6.11.2/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	SIP-I support. The called user is busy. Ensure that, when the called user in the PSTN/PLMN part of Network B, and ISUP – SIP-I interworking applied 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 --[any boundary name]--
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 486 Busy Here(REL) ACK →	
Comments	Establish a communication from Network A to Network B, user B in the PSTN/PLMN part of Network B is busy. Check: Is a 486 Busy Here sent from Network B to Network A? Check: Is a ISUP REL encapsulated and the Cause value indicator is set to '17'? Check: If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL? Repeat this test in reverse direction.

Test case number	SS_unsucc_014
Test case group	BCALL/unsuccessful
Reference	6.11.2/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	SIP-I support. The called user rejects the call. Ensure that, when the called user in the PSTN/PLMN part of Network B, and ISUP – SIP-I interworking applies in Network B, rejects the communication setup, the network initiates call clearing to the calling user with a 480 Temporarily Unavailable final response. 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 → ← 480 Temporarily Unavailable (REL) ACK →	
Comments	Establish a communication from Network A to Network B, user B in the PSTN/PLMN part of Network B rejects the communication setup. Check: Is a 480 Temporarily Unavailable sent from Network B to Network A Check: Is an ISUP REL encapsulated and is the Cause value indicator 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47
Test purpose	SIP-I support. Call clearing due to no answer from the called user initiated by the calling user Ensure that when the early dialogue is not confirmed by the called user, the calling user located in the PSTN/PLMN part of Network A, and ISUP – SIP-I interworking applies in Network A, initiates call clearing to the called user with CANCEL or BYE. An ISUP REL message is encapsulated in the BYE request and the Cause value indicator is set to '16'.
Configuration	
SIP Parameter	480: Reason: Q.850;cause=16 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required

	REL Cause value: 16 --[any boundary name]--

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

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

Test case number	SS_unsucc_016
Test case group	BCALL/unsuccessful
Reference	7.7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47
Test purpose	SIP-I support. Call clearing due to no answer from the called user initiated by the originating network. Ensure when the early dialogue is not confirmed by the called user, the originating network initiate the call clearing after timeout of ISUP timer T9 if the calling user is located in the PSTN/PLMN part of Network A, and ISUP – SIP-I interworking applies in Network A, and the originating network sends a CANCEL or BYE to the called user. An ISUP REL

	message is encapsulated in the BYE request and the Cause value indicator is set to '19'.
Configuration	
SIP Parameter	480: Reason: Q.850;cause=19 (optional) Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value: 19 --[any boundary name]--
Message flow SIP (Network A) Interconnection Interface SIP (Network B) → INVITE → ← 180 Ringing Start timer T9 Timeout T9 CASE A CANCEL → ← 200 OK CANCEL ← 487 Request Terminated ACK → CASE B BYE(REL) → ← 200 OK BYE(RLC) ← 487 Request Terminated ACK →	
Comments	Establish a communication from Network A to Network B, user B does not answer the communication setup. The ISUP timer T9 in the PSTN/PLMN expires Check: Is a CANCEL or BYE request is sent by the originating network? Check: Is 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/[ETSI TS 124 607]
SELECTION EXPRESSION	
Test purpose	No P-Preferred-Identity received. The terminating user receives the default public user identity of the originating user. If 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 case number	SS_oip_002
Test case group	SIP-SIP/Service/OIP
Reference	5.2.6.3/[ETSI TS 124 607]
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. If 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. If it 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 and 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 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/[ETSI TS 124 607]
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. If 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. If it 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 that 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 set to phone? Check: Is the P-Preferred-Identity header not present? Repeat this test in reverse direction. Repeat this test with all relevant end devices.

Test case number	SS_oip_004
Test case group	SIP-SIP/Service/OIP
Reference	4.5.2.4/[ETSI TS 124 607]
SELECTION EXPRESSION	SE 18 AND NOT SE 19
Test purpose	No Special arrangement exists. The special arrangement does not exist (screening of user provided information). The network compares the information in the From header with the set of registered public identities of the originating user. If no match is found, the AS sets the From header to the SIP URI that includes the registered default public user identity.
Configuration	Special arrangement for the originating user does not exist
SIP Parameter	INVITE From=default public user identity P-Asserted-Header=[any registered public user identity]
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ **Apply post test routine**	
Comments	Check: Is the From header URI set to the value of the P-Asserted-Identity URI? Check: Is the P-Asserted-Identity set to any registered public user identity? Check: Is the user parameter set to phone? Repeat this test in reverse direction. Repeat this test with all relevant devices.

Test case number	SS_oip_005
Test case group	SIP-SIP/Service/OIP
Reference	4.5.2.4/[ETSI TS 124 607]
SELECTION EXPRESSION	SE 18 AND SE 19
Test purpose	Special arrangement exists. The special arrangement exists (no screening of user provided information). The network does not attempt to match the information in the From header with the set of registered public identities of the originating user. The From header field is transparently transported to the terminating user.
Configuration	Special arrangement for the originating user exists
SIP Parameter	INVITE From= original value P-Asserted-Header=[any registered public user identity]
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ **Apply post test routine**	

Comments	Check: Is the From header URI set to original value sent by the user? Check: Is the P-Asserted-Identity set to any registered public user identity? Check: Is the user parameter set to phone? Repeat this test in reverse direction. Repeat this test with all relevant end devices.

Test case number	SS_oip_006
Test case group	SIP-SIP/Service/OIP
Reference	7.1.3/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 52
Test purpose	SIP-I support. ISUP Calling party number *presentation allowed* in the encapsulated IAM. Ensure when BICC/ISUP – SIP-I interworking applies in the originating network that 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 INVITE request? Check: Is the Calling party number present in the encapsulated IAM and is the screening indicator set to 'Network provided' or 'user provided, verified and passed' and is the Presentation restriction indicator set to 'allowed'? Check: Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM? Check: Is the value 'id' not present in the Privacy header field (if included)? Repeat this test in reverse direction.

Test case number	SS_oip_007
Test case group	SIP-SIP/Service/OIP
Reference	7.1.3/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 52 AND SE 52A
Test purpose	SIP-I support. ISUP Additional Calling party number *presentation allowed* in the encapsulated IAM. Ensure when BICC/ISUP – SIP-I interworking applies in the originating network that the BICC/ISUP IAM is encapsulated in the INVITE request. The From field is derived from the Additional Calling party number in the encapsulated IAM. The 'Presentation restriction' indicator in the encapsulated IAM is set to 'allowed'. No Privacy value 'id' is present in the INVITE request.
Configuration	The originating user in the PSTN/PLMN part of Network A is subscribed to the 'no screening option'
SIP Parameter	INVITE From=[derived from the ISUP Additional calling party number] P-Asserted-Identity=[derived from the ISUP calling party number] Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Calling party number Screening indicator Network Provided Presentation restriction allowed Address signal Generic number Number Qualifier Indicator Additional calling party number Screening indicator user provided, not verified Presentation restriction allowed Address signal

	--[any boundary name]--

Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(IAM) ➔ **Apply post test routine**

Comments	Check: Is a BICC/ISUP IAM encapsulated in the in the INVITE request? Check: Is the Calling party number present in the encapsulated IAM and is the screening indicator set to 'Network Provided' and is the Presentation restriction indicator set to 'allowed'? Check: Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM? Check: Is a Generic number parameter, Number Qualifier Indicator set to Additional calling party number present and is the screening indicator set to 'user provided, not verified' and is the Presentation restriction indicator set to 'allowed'? Check: Is the From header field derived from the Additional calling party number in the encapsulated IAM? Check: Is the value 'id' not present in the Privacy header field (if included)? Repeat this test in reverse direction.

7.1.5.2 Test purposes for OIR

Test case number	SS_oir_001
Test case group	SIP-SIP/Service/OIR
Reference	4.3.2, 4.5.2.4/[ETSI TS 124 607]
SELECTION EXPRESSION	SE 20
Test purpose	Terminating user does not receive the identity of the originating user. In case the preconditions are fulfilled not to provide the terminating UE with originating identification information (e.g., permanent mode), ensure that the P-Asserted-Identity still contains identity information and the privacy is set to 'id' or 'header' or 'user'. The terminating user does not receive the identity of the originating user. As a network option, the From header is set to an anonymous User Identity.
Configuration	Originating user subscribes to the OIR service
SIP Parameter	INVITE P-Asserted-Identity: Privacy:id OR header OR user From: <sip:anonymous@anonymous.invalid> (optional)

Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ **Apply post test routine**

Comments	Check: Is the P-Asserted-Identity is present? Check: Is the Privacy header set to 'id' or 'header' or 'user'? Check: Is (optional) the From header set to an anonymous User Identity? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_oir_002
Test case group	SIP-SIP/Service/OIR
Reference	4.3.2, 4.5.2.4/[ETSI TS 124 607]
SELECTION EXPRESSION	SE 20 AND SE 25
Test purpose	Communication forwarding unconditional, served user subscribes OIR. User A and user C are in Network B, and user C is provided with OIP. User B is in Network A and is provided with CFU "diverting number is released to the diverted-to user" = Yes. If the served user subscribes Originating Identification Restriction (e.g., permanent mode), ensure that when user A calls user B, the call is forwarded unconditional to user C, user C is not informed of the forwarding number. The diverted-to user receives no identity of the diverting user neither in a History-Info header nor in the To header.
Configuration	Diverting user subscribes to the OIR service
SIP Parameter	INVITE1: no history entry present INVITE2: History-Info: <sip:userB@networkA?Privacy=history >;index=1, <sip: userC@networkB;cause=302 >;index=1.1
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 escaped in the hi-targeted-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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 52
Test purpose	SIP-I support. ISUP Calling party number *presentation restricted* in the encapsulated IAM. Ensure when BICC/ISUP – SIP-I interworking applies in the originating network that 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 INVITE request? Check: Is the Calling party number present in the encapsulated IAM and is the screening indicator set to 'Network provided' or 'user provided, verified and passed' and is the Presentation restriction indicator set to 'restricted'? Check: Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM? Check: Is the value 'id' present in the Privacy header field? Repeat this test in reverse direction.

Test case number	SS_oir_004
Test case group	SIP-SIP/Service/OIR
Reference	7.1.3/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 52 AND 52A
Test purpose	SIP-I support. ISUP Additional Calling party number *presentation restricted* in the encapsulated IAM. Ensure that when BICC/ISUP – SIP-I interworking applies in the originating network, the BICC/ISUP IAM is encapsulated in the INVITE request. The From field is derived from the Additional Calling party number in the encapsulated IAM. The 'Presentation restriction' indicator in the Generic number parameter is set to 'allowed' no Privacy value 'id' is present in the INVITE request.
Configuration	The originating user in the PSTN/PLMN part of Network A is subscribed to the 'no screening option'.
SIP Parameter	INVITE P-Asserted-Identity=[derived from the ISUP calling party number] From=[derived from the ISUP Additional calling party number] Privacy: id Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Calling party number Screening indicator Network Provided Presentation restriction restricted Address signal Generic number Number Qualifier Indicator Additional calling party number Screening indicator user provided, not verified Presentation restriction restricted Address signal --[any boundary name]--
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(IAM) ➔ **Apply post test routine**	

Comments	Check: Is a BICC/ISUP IAM encapsulated in the INVITE request? Check: Is the Calling party number present in the encapsulated IAM, is the screening indicator set to 'Network Provided', and is the Presentation restriction indicator set to 'restricted'? Check: Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM? Check: Is a Generic number parameter, Number Qualifier Indicator set to Additional calling party number present and is the screening indicator set to 'user provided, not verified' and is the Presentation restriction indicator set to 'restricted' Check: Is the From header field derived from the Additional calling party number in the encapsulated IAM? Check: Is the value 'id' present in the Privacy header field? Repeat this test in reverse direction.

7.1.5.3 Test purposes for TIP

Test case number	SS_tip_001
Test case group	SIP-SIP/Service/TIP
Reference	5.2.6.4/[ETSI TS 124 608]
SELECTION EXPRESSION	
Test purpose	Originating user receives the identity of the terminating user. Ensure in case the preconditions are fulfilled to provide the originating UE with terminating identification information without preventing the presentation, the originating UE receives in a 1xx or 200 SIP response, a P-Asserted-Identity header field with a valid public user identity of the terminating UE.
Configuration	
SIP Parameter	18x/200 OK INVITE P-Asserted-Identity:
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE → CASE A ← 180 Ringing CASE B ← 183 Session Progress CASE C ← 200 OK INVITE(P-Asserted-Identity) Apply post test routine	
Comments	Check: Is the P-Asserted-Identity is present in a 180 Ringing or 183 Session Progress or in a 200 OK INVITE? Repeat this test in reverse direction. Repeat this test with all relevant end devices.

Test case number	SS_tip_002
Test case group	SIP-SIP/Service/TIP
Reference	4.5.2.9/[ETSI TS 124 608]
SELECTION EXPRESSION	SE 21 AND SE 22 AND [Network B] SE 48
Test purpose	Second identity provided in UPDATE. Ensure that, when the option tag "from-change" in the Supported header field is provided by the originating UE in the INVITE request and the terminating UE receives the from-change tag, the terminating user sends a 'from-change' tag in the supported header in the 200 OK INVITE. A second identity is provided in the UPDATE request sent by the terminated user in the From header after the ACK is 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: Does an UPDATE request sent by the terminating user contain a From header field set to the value sent by the terminating user? Repeat this test in reverse direction. Repeat this test with all chosen end devices.

Test case number	SS_tip_003
Test case group	SIP-SIP/Service/TIP
Reference	4.5.2.9/[ETSI TS 124 608]
SELECTION EXPRESSION	SE 21 AND SE 22 AND [Network B] SE 48
Test purpose	Second identity not provided

	Ensure that, when the option tag "from-change" in the Supported header field is provided by the originating UE in the INVITE request, the terminating user does not receive the from-change tag in the initial INVITE, no from-change tag is sent in the 200 OK INVITE response, an UPDATE containing a second identity is sent and the From header is set to the default public user identity of the terminating user.
Configuration	Special arrangement for the terminating user does not exist
SIP Parameter	INVITE Supported: from-change 200 OK INVITE P-Asserted-Identity: UPDATE From: (default public user identity)
Message flow SIP (Network A)	Interconnection Interface INVITE ➔ 🡐 180 Ringing 🡐 200 OK INVITE(P-Asserted-Identity) ACK ➔ 🡐 UPDATE (From) 200 OK UPDATE ➔ Apply post test routine SIP (Network B)
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: Does an UPDATE request sent by the terminating user contain a From header field set to the public user identity of the terminating user? Repeat this test in reverse direction. Repeat this test with all relevant end devices.

Test case number	SS_tip_004
Test case group	SIP-SIP/Service/TIP
Reference	6.7/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 53
Test purpose	SIP-I support. The Connected number presentation allowed is present in the encapsulated 200 OK. Ensure that on receipt of a 200 OK INVITE to establish a confirmed dialogue, an ANM is encapsulated if SIP-I – BICC/ISUP interworking is applicable in Network B. The Address presentation restriction indicator is set to 'allowed'. The screening indicator is set to Network provided or user provided, verified and passed.
Configuration	

SIP Parameter	200 OK INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM Connected number Screening indicator Network provided or user provided, verified and passed Address presentation restriction allowed Address signal --[any boundary name]--
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(IAM) → ← 180 Ringing(ACM) ← 200 OK INVITE(ANM) ACK → Apply post test routine	
Comments	Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response? Check: Is the Screening indicator in the encapsulated ANM set to 'Network provided' or 'user provided, verified and passed'? Check: Is the Address presentation restriction indicator in the encapsulated ANM set to 'allowed'? Repeat this test in reverse direction.

Test case number	SS_tip_005
Test case group	SIP-SIP/Service/TIP
Reference	6.7/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 53 AND SE 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 'Network provided' and the Address Presentation Restricted indicator is set to 'allowed'.

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

7.1.5.4 Test purposes for TIR

Test case number	SS_tir_001
Test case group	SIP-SIP/Service/TIR
Reference	4.5.2.9/[ETSI TS 124 608]
SELECTION EXPRESSION	SE 23
Test purpose	Originating user does not receive the identity of the terminating user. Ensure that when the preconditions are fulfilled to prevent the presentation of the terminating user identity at the originating user, the originating UE receives, in any non-100 SIP response (e.g., 180, 183, 200), a Privacy header field set to "id". 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 present in the provisional (18x) or final (200 OK) response? Check: Is the Privacy header in the provisional (18x) or final (200 OK) response 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/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 23
Test purpose	CDIV occurs. Originating user does not receive the identity of the served user. Ensure that, when Call diversion occurs, the identity of the CDIV served user is restricted when the CDIV served user is subscribed to the TIR service and requires to prevent the presentation of his/here identity.

	The hi-entry of the History-Info header in the 181 identifying the served user contains an escaped 'Privacy' header set to 'history'.
Configuration	The served user subscribes to the 'TIR' service
SIP Parameter	181 History-Info: <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@networkB;cause=[any] >;index=1.1
Message flow SIP (Network A) Apply post test routine	 Interconnection Interface — SIP (Network B) INVITE ➔ 🡐 181 Being Forwarded 🡐 INVITE
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_001B
Test case group	SIP-SIP/Service/TIR
Reference	4.5.2.7/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 23
Test purpose	CDIV occurs. Originating user does not receive the identity of the diverted to user. Ensure that, when Call diversion occurs, the identity of the diverted-to user is restricted when the diverted-to user is subscribed to the TIR service and requires to prevent the presentation of his/here identity. The hi-entry of the History-Info header in the 180 or 200 OK INVITE identifying the diverted-to user contains an escaped 'Privacy' header set to 'history'.
Configuration	The diverted-to user subscribes to the 'TIR' service
SIP Parameter	180/200 OK History-Info: <sip:userB@networkB>;index=1, <sip: userC@networkB;cause=[any]?Privacy=history>;index=1.1
Message flow SIP (Network A)	 Interconnection Interface — SIP (Network B) INVITE(1) ➔ 🡐 INVITE(2) 180 Ringing(2) ➔ 🡐 180 Ringing(1) 200 OK INVITE(2) ➔ 🡐 ACK

← 200 OK INVITE(1) ACK → Apply post test routine	
Comments	Check: Is the History-Info header present in the 180 or 200 OK sent to the originating user? Check: 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 this test with all chosen end devices.

Test case number	SS_tir_002
Test case group	SIP-SIP/Service/TIR
Reference	6.7/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 53
Test purpose	SIP-I support. The Connected number presentation allowed is present in the encapsulated 200 OK. Ensure that on receipt of a 200 OK INVITE to establish a confirmed dialogue an ANM is encapsulated if SIP-I – BICC/ISUP interworking is applicable in Network B. The Address presentation restriction indicator is set to 'restricted'. The screening indicator is set to 'Network provided' or 'user provided, verified and passed'.
Configuration	
SIP Parameter	200 OK INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM Connected number Screening indicator Network provided or user provided, verified and passed Address presentation restriction restricted Address signal --[any boundary name]--
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(IAM) → ← 180 Ringing(ACM) ← 200 OK INVITE(ANM) ACK → Apply post test routine

Comments	Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response? Check: Is the Screening indicator in the encapsulated ANM set to 'Network provided' or 'user provided, verified and passed'? Check: Is the Address presentation restriction indicator in the encapsulated ANM set to 'allowed'? Repeat this test in reverse direction.

Test case number	SS_tir_003
Test case group	SIP-SIP/Service/TIR
Reference	6.7/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 53 AND SE 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 'restricted'. A Connected number parameter is present the Screening indicator is set to 'Network provided' and the Address Presentation Restricted indicator is set to 'restricted'.
Configuration	The terminating user in the PSTN/PLMN part of Network B is subscribed to the COLP 'no screening option'
SIP Parameter	200 OK INVITE P-Asserted-Identity=[derived from the ISUP Connected number] Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM Connected number Screening indicator Network provided or user provided, verified and passed Presentation restriction restricted Address signal Generic number Number Qualifier Indicator Additional 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	
Comments	Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response? Check: Is a Generic number parameter present in the encapsulated ANM? Check: Is the Number Qualifier Indicator of the Generic number set to 'additional connected number'? Check: Is the Screening indicator of the Generic number set to 'user provided, not verified'? Check: Is the Address presentation restriction indicator in the Generic number set to 'allowed'? Repeat this test in reverse direction.

7.1.5.5 Communication hold (HOLD)

Test case number	SS_hold_001
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[ETSI TS 124 610]
SELECTION EXPRESSION	SE 24
Test purpose	Hold the session where the media stream was previously set to sendrecv. Ensure that the UE A requesting hold of the session sends an INVITE or UPDATE request to hold the session. Hold is done containing the SDP with the attribute "a=sendonly". The UE A, after requesting the hold session, *receives* 200 OK final response containing the SDP with the attribute "a=recvonly".
Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) A confirmed session already exists CASE A INVITE(sendonly) → ← 200 OK INVITE (recvonly) ACK → CASE B UPDATE(sendonly) → ← 200 OK UPDATE (recvonly) Apply post test routine	
Comments	Check: Is the user in Network A able to set the session on hold by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented?

	Repeat this test in reverse direction.

Test case number	SS_hold_002
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[ETSI TS 124 610]
SELECTION EXPRESSION	SE 24
Test purpose	Hold the session where the media stream was previously set to recvonly. Ensure that the UE B requesting hold of the session stops sending media and sends an INVITE or UPDATE request to hold the session. Hold is done containing the SDP with the attribute "a=sendonly". The UE A, after requesting to hold the held session, sends an INVITE or UPDATE request containing the SDP with the attribute "a=inactive."
Configuration	
SIP Parameter	

Message flow				
SIP (Network A)		Interconnection Interface		SIP (Network B)
		A confirmed session already exists		
CASE A	🡐	INVITE (sendonly)		
		200 OK INVITE (recvonly)	🡒	
	🡐	ACK		
		INVITE (inactive)	🡒	
	🡐	200 OK INVITE (inactive)		
		ACK	🡒	
CASE B	🡐	INVITE (sendonly)		
		200 OK INVITE (recvonly)	🡒	
	🡐	ACK		
		UPDATE(inactive)	🡒	
	🡐	200 OK UPDATE (inactive)		
CASE C	🡐	UPDATE (sendonly)		
		200 OK UPDATE (recvonly)	🡒	
		INVITE (inactive)	🡒	
	🡐	200 OK INVITE (inactive)		
		ACK	🡒	
CASE D	🡐	UPDATE (sendonly)		
		200 OK UPDATE (recvonly)	🡒	
		UPDATE(inactive)	🡒	
	🡐	200 OK UPDATE (inactive)		
		Apply post test routine		

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

Test case number	SS_hold_003
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[ETSI TS 124 610]
SELECTION EXPRESSION	SE 24
Test purpose	Resume the session where the media stream was previously set to sendonly. Ensure that when the UE A is requested to resume the session with user B, the UE-A starts sending media and sends an INVITE or UPDATE request to resume the session with the attribute "a=sendrecv" in the SDP. The UE A, after requesting to resume the held session, *receives* 200 OK final response and optionally the attribute "a=sendrecv" in the SDP. The a=sendrecv attribute is the default value therefore the attribute can be omitted.
Configuration	
SIP Parameter	

Message flow

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

CASE D	UPDATE (sendonly) → ← 200 OK UPDATE (recvonly) UPDATE (sendrecv) → ← 200 OK UPDATE (sendrecv) Apply post test routine
Comments	Check: Is the user in Network A able to set the session on hold by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented? Check: Is the user in Network A able to retrieve the session by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented? The absence of the 'sendrecv' attribute is the default value. Repeat this test in reverse direction.

Test case number	SS_hold_004
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[ETSI TS 124 610]
SELECTION EXPRESSION	SE 24
Test purpose	Resume the session where the media stream was previously set to inactive. The Session is in the "inactive" state. Ensure that when the UE A is requesting to resume the session with user B, the UE-A sends an INVITE or UPDATE to resume the session with the attribute "a=recvonly" in the SDP. The UE A after requesting to resume the held session *receives* 200 OK final response with the attribute "a=sendonly" in the SDP.
Configuration	
SIP Parameter	
Message flow	

SIP (Network A)		Interconnection Interface		SIP (Network B)
		A confirmed session already exists		
CASE A	←	INVITE(sendonly)		
		200 OK INVITE (recvonly)	→	
	←	ACK		
		INVITE(inactive)	→	
	←	200 OK INVITE (inactive)		
		ACK	→	
		INVITE (recvonly)	→	
	←	200 OK INVITE (sendonly)		
		ACK	→	
CASE B	←	INVITE(sendonly)		
		200 OK INVITE (recvonly)	→	
	←	ACK		
		UPDATE(inactive)	→	
	←	200 OK UPDATE (inactive)		

	INVITE (recvonly) → ← 200 OK INVITE (sendonly) ACK → CASE C ← UPDATE (sendonly) 200 OK UPDATE (recvonly) → INVITE(inactive) → ← 200 OK INVITE (inactive) ACK → UPDATE (recvonly) → ← 200 OK UPDATE (sendonly) CASE D ← UPDATE (sendonly) 200 OK UPDATE (recvonly) → UPDATE(inactive) → ← 200 OK UPDATE (inactive) UPDATE (recvonly) → ← 200 OK UPDATE (sendonly) Apply post test routine
Comments	Check: Is the user in Network B able to set the session on hold by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented? Check: Is the user in Network A able to set the session on hold by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented? Check: Is the user in Network A able to retrieve the session by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented? Repeat this test in reverse direction.

Test case number	SS_hold_005
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[ETSI TS 124 610]
SELECTION EXPRESSION	SE 24
Test purpose	Hold the session the media stream was previously set at to sendrecv. Ensure that the UE B sends an INVITE or UPDATE request to hold the session. Hold is done containing the SDP with the attribute "a=sendonly". The UE A *sends* a 200 OK final response containing the SDP with the attribute "a=recvonly" and stops sending media.
Configuration	
SIP Parameter	

Message flow

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

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

Test case number	SS_hold_006
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[ETSI TS 124 610]
SELECTION EXPRESSION	SE 24
Test purpose	Hold the session the media stream was previously set at to sendonly. The Session is in the held state done by UE-A. Ensure that the UE B sends an INVITE or UPDATE request to hold the session. Hold is done containing the SDP with the attribute "a=inactive". The UE A after receiving the hold *sends* 200 OK final response containing the SDP with the attribute "a=inactive" and stops sending media.
Configuration	
SIP Parameter	

Message flow

SIP (Network A)		Interconnection Interface		SIP (Network B)
		A confirmed session already exists		
CASE A		INVITE(sendonly)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	INVITE (inactive)		
		200 OK INVITE (inactive)	→	
	←	ACK		
CASE B		INVITE(sendonly)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	UPDATE (inactive)		
		200 OK UPDATE (inactive)	→	

CASE C	UPDATE (sendonly) →	
	← 200 OK UPDATE (recvonly)	
	← INVITE (inactive)	
	200 OK INVITE (inactive) →	
	← ACK	
CASE D	UPDATE (sendonly) →	
	← 200 OK UPDATE (recvonly)	
	← UPDATE (inactive)	
	200 OK UPDATE (inactive) →	
	Apply post test routine	
Comments	Check: Is the user in Network A able to set the session on hold by sending an INVITE or UPDATE request 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 is the version parameter in the SDP 'o' line incremented? Repeat this test in reverse direction.	

Test case number	SS_hold_007
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[ETSI TS 124 610]
SELECTION EXPRESSION	SE 24
Test purpose	Resume the session the media stream was previously set at to recvonly. Ensure that the UE B sends an INVITE or UPDATE request requesting to resume the session with user A, the UE-B starts sending media. Resume is done containing the SDP with the attribute "a=sendrecv". The UE A after receiving the Resume of the session *sends* 200 OK final response containing the SDP with the attribute "a=sendrecv". The a=sendrecv attribute is the default value therefore the attribute can be omitted.
Configuration	
SIP Parameter	

Message flow

SIP (Network A)	Interconnection Interface	SIP (Network B)
	A confirmed session already exists	
CASE A	← INVITE (sendonly)	
	200 OK INVITE(recvonly) →	
	← ACK	
	← INVITE(sendrecv)	
	200 OK INVITE(sendrecv) →	
	← ACK	
CASE B	← UPDATE (sendonly)	
	200 OK UPDATE (recvonly) →	

← UPDATE (sendrecv) 200 OK UPDATE (sendrecv) → Apply post test routine	
Comments	Check: Is the user in Network B able to set the session on hold by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented? Check: Is the user in Network B able to retrieve the session by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented? Repeat this test in reverse direction.

Test case number	SS_hold_008
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[ETSI TS 124 610]
SELECTION EXPRESSION	SE 24
Test purpose	Resume the session where the media stream was previously set to inactive. The Session is in the "inactive" state. Ensure that the UE B sends an INVITE or UPDATE request requesting to resume the session with user B, the UE-A starts sending media. Resume is done containing the SDP with the attribute "a=recvonly". The UE A after receiving the Resume of the session *sends* 200 OK final response containing the SDP with the attribute "a=sendonly". The a=sendrecv attribute is the default value therefore the attribute can be omitted.
Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) A confirmed session already exists CASE A INVITE (sendonly) → ← 200 OK INVITE (recvonly) ACK → ← INVITE (inactive) 200 OK INVITE (inactive) → ← ACK ← INVITE (recvonly) 200 OK INVITE (sendonly) → ← ACK CASE B INVITE (sendonly) → ← 200 OK INVITE (recvonly) ACK → ← UPDATE (inactive) 200 OK UPDATE (inactive) → ← UPDATE (recvonly) 200 OK UPDATE (sendonly) →	

CASE C UPDATE (sendonly) → ← 200 OK UPDATE (recvonly) ← INVITE (inactive) 200 OK INVITE (inactive) → ← ACK ← INVITE (recvonly) 200 OK INVITE (sendonly) → ← ACK CASE D UPDATE (sendonly) → ← 200 OK UPDATE (recvonly) ← UPDATE (inactive) 200 OK UPDATE (inactive) → ← UPDATE (recvonly) 200 OK UPDATE (sendonly) → Apply post test routine	
Comments	Check: Is the user in network 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 is the version parameter in the SDP 'o' line incremented? Check: Is the user in Network B able to retrieve the session by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented? Repeat this test in reverse direction.

Test case number	SS_hold_009
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[ETSI TS 124 610]
SELECTION EXPRESSION	SE 24
Test purpose	Resume the session on both sides where the media stream was previously set to inactive. The Session is in the "inactive" state. Ensure that the UE A is requesting to resume the session with user B, the UE-A starts sending media and sends an INVITE or UPDATE request to resume the session with the attribute "a=recvonly in the SDP. The UE A after requests to resume the session *receives* 200 OK final response containing the SDP with the attribute "a=sendonly. The UE B after requests to resume the session *receives* 200 OK final response containing the SDP with the attribute "a=sendrecv". The a=sendrecv attribute is the default value therefore the attribute can be omitted.

Message flow				
SIP (Network A)		Interconnection Interface		SIP (Network B)
		A confirmed session already exists		
CASE A		INVITE(sendonly)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	INVITE(inactive)		
		200 OK INVITE (inactive)	→	
	←	ACK		
		INVITE(recvonly)	→	
	←	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)		
		ACK	→	
	←	UPDATE (sendrecv)		
		200 OK UPDATE (sendrecv)	→	
CASE C		UPDATE (sendonly)	→	
	←	200 OK UPDATE (recvonly)		
	←	INVITE(inactive)		
		200 OK INVITE (inactive)	→	
	←	ACK		
		UPDATE (recvonly)	→	
	←	200 OK UPDATE (sendonly)		
		ACK	→	
	←	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) 200 OK UPDATE (sendrecv) → Apply post test routine	
Comments	Check: Is the user in Network A able to set the session on hold by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented? Check: Is the user in Network B able to set the session on hold by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented? Check: Is the user in Network A able to retrieve is the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line incremented? Check: Is the user in Network B able to retrieve the session by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented? The absence of the 'sendrecv' attribute is the default value. Repeat this test in reverse direction.

Test case number	SS_hold_010
Test case group	SIP-SIP/Service/HOLD
Reference	4.5.2.1/[ETSI TS 124 610]
SELECTION EXPRESSION	SE 24
Test purpose	Resume the session on both sides where the media stream was previously set to inactive. The Session is in the "inactive" state. Ensure that the UE B sends an INVITE or UPDATE request to resume the session with user A, the UE-B starts sending media. Resume is done containing the SDP with the attribute "a=recvonly". The UE A after receiving the Resume of the session *sends* 200 OK final response containing the SDP with the attribute "a=sendonly". The UE A after requests to resume the session sends an INVITE or UPDATE requestcontaining the SDP with the attribute "a=sendrecv. The UE B after receiving the Resume of the session *sends* 200 OK final response containing the SDP with the attribute "a=sendrecv". The a=sendrecv attribute is the default value therefore the attribute can be omitted.
Configuration	
SIP Parameter	
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) A confirmed session already exists CASE A ← INVITE(sendonly) 200 OK INVITE (recvonly) → ← ACK INVITE(inactive) → ← 200 OK INVITE (inactive)	

		ACK	→
	←	INVITE(recvonly)	
		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)	→
	←	ACK	
		UPDATE (sendrecv)	→
	←	200 OK UPDATE (sendrecv)	
CASE C	←	UPDATE (sendonly)	
		200 OK UPDATE (recvonly)	→
		INVITE(inactive)	→
	←	200 OK INVITE (inactive)	
		ACK	→
	←	UPDATE (sendonly)	
		200 OK UPDATE (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)	→
	←	200 OK UPDATE (sendrecv)	
		Apply post test routine	
Comments	Check: Is the user in Network B able to set the session on hold by sending an INVITE or UPDATE request and is the version parameter in the SDP 'o' line incremented?		

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

Test case number	SS_hold_011
Test case group	SIP-SIP/Service/HOLD
Reference	B.10/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 54
Test purpose	SIP-I support. Hold requested by the calling user. Ensure that when an INVITE request updates a confirmed session a CPG is encapsulated if ISUP – SIP-I interworking is applicable in Network A. The Generic Notification Indicator parameter is present set to 'hold'. The 'a' attribute is set to 'sendonly' present in the SDP. In the 200 OK INVITE the 'a' attribute is set to 'recvonly' present in the SDP.
Configuration	
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] a=sendonly --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic notification remote hold --[any boundary name]--
Message flow SIP (Network A) Interconnection Interface SIP (Network B) A confirmed session already exists CASE A INVITE(sendonly, CPG hold) → ← 200 OK INVITE (recvonly) ACK → Apply post test routine	
Comments	Establish a session from Network A to Network B

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

Test case number	SS_hold_012
Test case group	SIP-SIP/Service/HOLD
Reference	B.10/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 54
Test purpose	SIP-I support. Hold requested by the called user. Ensure that when an INVITE request updates a confirmed session, a CPG is encapsulated if SIP-I – ISUP interworking is applicable in Network B. The Generic Notification Indicator parameter is present set to 'hold'. The 'a' attribute is set to 'sendonly' present in the SDP. In the 200 OK INVITE the 'a' attribute is set to 'recvonly' present in the SDP.
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] a=sendonly --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic notification remote hold --[any boundary name]--
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) A confirmed session already exists CASE A ← INVITE(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 in 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 54
Test purpose	SIP-I support. Hold requested by the originating user, Hold by the terminating user. Retrieve requested by the originating user. Ensure the hold and retrieve procedure when ISUP – SIP-I interworking applies in the Network A. • Originating user in Network A places the session on hold. • Terminating user in Network B places the session on hold. • Originating user in Network A retrieves the session. • Terminating user in Network B retrieves the session. Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A.
Configuration	
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic notification remote hold or remote retrieval --[any boundary name]--

Message flow				
SIP (Network A)		Interconnection Interface		SIP (Network B)
		A confirmed session already exists		
CASE A		INVITE(sendonly, CPG hold)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	
	←	INVITE(inactive)		
		200 OK INVITE (inactive)	→	
	←	ACK		
		INVITE(sendonly, CPG retrieval)	→	
	←	200 OK INVITE (recvonly)		
		ACK	→	

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

Test case number	SS_hold_014
Test case group	SIP-SIP/Service/HOLD
Reference	B.10/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 54
Test purpose	SIP-I support. Hold requested by the originating user, Hold by the terminating user. Retrieve requested by the terminating user. Ensure the hold and retrieve procedure when ISUP – SIP-I interworking applies in the Network A. • Originating user in Network A places the session on hold. • Terminating user in Network B places the session on hold. • Terminating user in Network B retrieves the session. • Originating user in Network A retrieves the session. Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A.
Configuration	
SIP Parameter	INVITE: Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic notification

	remote hold or remote retrieval --[any boundary name]--

Message flow

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

Comments	Establish a session from Network A to Network B. The user in the PSTN/PLMN part of Network A places the session on hold. Check: Is a CPG encapsulated in the INVITE request? Check: Is a Generic notification parameter present in 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 in 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 54
Test purpose	SIP-I support. Hold requested by the terminating user, Hold by the originating user. Retrieve requested by the originating user. Ensure the hold and retrieve procedure when ISUP – SIP-I interworking applies in the Network A. • Terminating user in Network B places the session on hold. • Originating user in Network A places the session on hold. • Originating user in Network A retrieves the session. • Terminating user in Network B retrieves the session. Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A.
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(inactive, CPG hold) → ← 200 OK INVITE (inactive) ACK → INVITE(recvonly, CPG retrieval) → ← 200 OK INVITE (sendonly) ACK → ← INVITE(sendrecv) 200 OK INVITE (sendrecv) → ← ACK Apply post test routine	
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 in 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 in the Notification indicator set to 'remote retrieval'? Check: Is the 'a' attribute in the SDP set to 'recvonly'? Check: Is the Version parameter in the SDP incremented? The user in Network B retrieves the session. Check: Is the 'a' attribute in the SDP set to 'sendrecv'? Check: Is the Version parameter in the SDP incremented? Repeat this test in reverse direction.

Test case number	SS_hold_016
Test case group	SIP-SIP/Service/HOLD
Reference	B.10/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 54
Test purpose	SIP-I support. Hold requested by the terminating user, Hold by the originating user. Retrieve requested by the terminating user Ensure the hold and retrieve procedure when ISUP – SIP-I interworking applies in the Network A • Terminating user in Network B places the session on hold. • Originating user in Network A places the session on hold. • Terminating user in Network B retrieves the session. • Originating user in Network A retrieves the session. Verify the Generic notification parameter in the encapsulated CPG present in the 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(inactive, CPG hold) → ← 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 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 in 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 in 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	SIP-SIP/Service/CFU
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 25
Test purpose	Communication forwarding unconditional, basic rules. User A and user C are in Network A. User B is in Network B and is provided with CFU.

	Ensure that when user A calls user B, the call is forwarded unconditional to user C. In the active call state, ensure the property of speech.
Configuration	
SIP Parameter	
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → CFU is performed ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ← ACK(Call-ID B-C) ← 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → Communication Apply post test routine	
Comments	Check: CDIV unconditional is successful Check: In the active call state, ensure the property of speech Check: Is the P-Asserted-Identity present in the INVITE sent from Network B to Network A set to the identity of the originating user? Repeat this test in reverse direction.

Test case number	SS_cfu_002
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	Communication forwarding unconditional, no notification. User A and user C are in Network A. 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, and the originating user is not notified.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = No
SIP Parameter	
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → CFU is performed ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) →	

	🡐 180 Ringing(Call-ID B-A) Apply post test routine
Comments	Check: No notification regarding call forwarding in Network B is received at the interconnection interface. Repeat this test in reverse direction.

Test case number	SS_cfu_003
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	Communication forwarding unconditional, originating user is notified. URI of the diverted-to user not received. User A and user C are in Network A. User B is in Network B and is provided with CFU. Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and not informed of the diverted-to number and served user number.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No
SIP Parameter	181 Being Forwarded P-Asserted-Identity: <userB@NetworkB> Privacy: id History-Info: <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@networkA;cause=302?Privacy=history>;index=1.1
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) 🡒 CFU is performed 🡐 INVITE(Call-ID B-C) 🡐 181 Being Forwarded(Call-ID B-A) Apply post test routine	
Comments	Check: A 181 Being Forwarded and a History-Info header is received at the interconnection interface in both entries. In the History-Info header a Privacy header is escaped value 'history' Check: Is the cause parameter in the last entry 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 this test in reverse direction.

Test case number	SS_cfu_004
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	Communication forwarding unconditional, originating user is notified. URI from the diverted-to user received. User A and user C are in Network 1. User B is in Network N2 and is provided with CFU. Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes. Ensure that when user A calls user B, the call is forwarded unconditional to user C, and 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> History-Info: *<sip:userB@networkB>;index=1*, <sip: userC@networkA;cause=302>;index=1.1
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFU is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) Apply post test routine
Comments	Check: A 181 Being Forwarded is received at the interconnection interface. Check: A History-Info header is contained in the 181 with the URI of the diverted-to user. Check: Is the cause parameter in the last entry 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 this test in reverse direction.

Test case number	SS_cfu_005
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	Communication forwarding unconditional, diverted-to user does not receive the URI of the served user. User A and user C are in Network A. User B is in Network B and is provided with CFU "Served user allows the presentation of his/her URI to the diverted-to user"= No. Ensure that when user A calls user B, the call is forwarded unconditional to user C, and user C is not informed of the forwarding number.
Configuration	Subscription options: • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE: History-Info: <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@networkA;cause=302>;index=1.1
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → CFU is performed ← INVITE(Call-ID B-C) Apply post test routine	
Comments	Check: A History-Info header 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 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 this test in reverse direction.

Test case number	SS_cfu_006
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	Communication forwarding unconditional, diverted-to user receives the URI of the served user.

	User A and user C are in Network A. User B is in Network B and is provided with CFU "Served user allows the presentation of his/her URI to diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded unconditional to user C, and user C is informed of the forwarding number.
Configuration	Subscription options: • Served user allows the presentation of his/her URI to diverted-to user = Yes
SIP Parameter	INVITE: History-Info: <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=302>;index=1.1
Message flow	SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → CFU is performed ← INVITE(Call-ID B-C) Apply post test routine
Comments	Check: A History-Info header is received in the INVITE and 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 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_cfu_007
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 25 AND SE 30
Test purpose	Communication forwarding unconditional, full notification. User A and user C are in Network A. User B is in Network B and is provided with CFU. Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, and "Served user allows the presentation of his/her URI to diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded unconditional to user C, and user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes

<table>
<tr><td></td><td>• Served user allows the presentation of his/her URI to diverted-to user = Yes</td></tr>
<tr><td>SIP Parameter</td><td>INVITE:
History-Info:
<sip:userB@networkB>;index=1,
<sip: userC@networkA;cause=302>;index=1.1

181 Being Forwarded
P-Asserted-Identity: <userB@NetworkB>
History-Info:
<sip:userB@networkB>;index=1,
<sip: userC@networkA;cause=302>;index=1.1

200 OK INVITE
History-Info header:
<sip:userB@networkB>;index=1,
<sip: userC@networkA;cause=302>;index=1.1</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
INVITE(Call-ID A-B) →
CFU is performed
← INVITE(Call-ID B-C)
← 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</td></tr>
<tr><td>Comments</td><td>Check: A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user.
Check: A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user.
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.</td></tr>
</table>

<table>
<tr><td>Test case number</td><td>SS_cfu_008</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CFU</td></tr>
<tr><td>Reference</td><td>4.5.2.6/[ETSI TS 124 604]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>SE 25</td></tr>
<tr><td>Test purpose</td><td>Communication forwarding unconditional, unsuccessful UDUB.
User A and user C are in Network A. User B is in Network B and is provided with CFU.
Ensure that when user A calls user B, the call is forwarded unconditional to user C and user C is user determined user busy</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td></td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(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) →</td></tr>
<tr><td>Comments</td><td>Check: The dialogue is terminated by receiving a 486 Busy Here
Repeat this test in reverse direction.</td></tr>
</table>

<table>
<tr><td>Test case number</td><td>SS_cfu_009</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CFU</td></tr>
<tr><td>Reference</td><td>4.5.2.6/[ETSI TS 124 604]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>SE 25</td></tr>
<tr><td>Test purpose</td><td>Communication forwarding unconditional, unsuccessful NDUB.
User A and user C are in Network A. 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.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td></td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(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) →</td></tr>
</table>

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

Test case number	SS_cfu_010
Test case group	SIP-SIP/Service/CFU
Reference	4.5.2.6/[ETSI TS 124 604
SELECTION EXPRESSION	SE 25 AND SE 30 AND [Network A] SE 9
Test purpose	Communication forwarding unconditional, interaction with a non trusted network. User A and user C are in Network A. Network A is non trusted. 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, and user A is notified of call diversion and not informed of the diverted-to number, and user C is not informed of the forwarding number.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFU is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) Apply post test routine
Comments	Check: No History-Info header is received in the INVITE at the interconnection interface. Check: No History-Info header is received in the 181 Being Forwarded at the interconnection interface (if sent). Repeat this test in reverse direction.

Test case number	SS_cfu_011
Test case group	SIP-SIP/Service/CFU
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	SIP-I support. CFU performed in Network B, Notification subscription options is set to presentation not allowed. User A and user C are in Network A. 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 = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded unconditional to user C, and user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = no
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation not allowed Redirecting reason unconditional Generic notification call is diverting --[any boundary name]--
Message flow	

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

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A Check: Is a 183 Session Progress received at the interconnection interface Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator is set to 'presentation not allowed'? Check: Is the Redirecting reason set to 'unconditional'? Repeat this test in reverse direction.

Test case number	SS_cfu_012
Test case group	SIP-SIP/Service/CFU
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	SIP-I support. CFU performed in Network B, Notification subscription options is set to presentation allowed without redirection number. User A and user C are in Network A. 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 = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded unconditional to user C, and user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number

SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed without redirection number Redirecting reason unconditional Generic notification call is diverting --[any boundary name]--
Message flow SIP (Network A) | Interconnection Interface | SIP (Network B) INVITE(Call-ID A-B) → CFU is performed ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: 183 Session Progress is received at the interconnection interface. Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator set to 'presentation allowed without redirection number'? Check: Is the Redirecting reason set to 'unconditional'? Repeat this test in reverse direction.

Test case number	SS_cfu_013
Test case group	SIP-SIP/Service/CFU
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	SIP-I support. CFU performed in Network B, Notification subscription options is set to presentation allowed with redirection number. User A and user C are in Network A. 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 = yes, with diverted-to user number. Ensure that when user A calls user B, the call is forwarded unconditional to user C, and user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number.
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed with redirection number Redirecting reason unconditional Generic notification call is diverting --[any boundary name]--

Message flow	
SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFU is performed ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: 183 Session Progress is received at the interconnection interface. Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator set to 'presentation allowed 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	SIP-I support. CFU performed in Network B, Restriction of the Redirection number. User A and user C are in Network A. 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, and a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	Subscription options: • Connected user subscribed to COLR, Permanent = yes
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM Redirection number restriction Presentation restricted --[any boundary name]--

Message flow		
SIP (Network A)	Interconnection Interface	SIP (Network B)
	INVITE(Call-ID A-B) →	
	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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53
Test purpose	SIP-I support. CFU performed in Network B, No restriction of the Redirection number. User A and user C are in Network A. 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, and 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

<table>
<tr><td>SIP Parameter</td><td>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]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →
CFU is performed
← INVITE(Call-ID B-C, IAM)
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</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A.
Check: Is a 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.</td></tr>
</table>

Test case number	SS_cfu_016
Test case group	SIP-SIP/Service/CFU
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	SIP-I support. CFU performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'. User A and user C are in Network A. 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, and user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC – SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason unconditional --[any boundary name]--

Message flow			
SIP (Network A)	Interconnection Interface		SIP (Network B)
	INVITE(Call-ID A-B)	➔	
	CFU is performed		
🡐	INVITE(Call-ID B-C, IAM)		
	Apply post test routine		

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and is the Address presentation restricted indicator set to 'presentation allowed'? Check: Is the Original called number present and is the Address presentation restricted indicator set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is Redirection information present and is the Redirecting reason set to 'unconditional'? Repeat this test in reverse direction.

Test case number	SS_cfu_017
Test case group	SIP-SIP/Service/CFU
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55
Test purpose	SIP-I support. CFU performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted' User A and user C are in Network A. 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, and user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if ISUP/BICC – SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Do not release diverting number information

SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation restricted Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason unconditional --[any boundary name]--
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) ➔ CFU is performed 🡐 INVITE(Call-ID B-C, IAM) Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and is the Address presentation restricted indicator set to 'presentation restricted'? Check: Is the Original called number present and is the Address presentation restricted indicator set to 'presentation restricted'? Check: Is the Redirection number present? Check: Is Redirection information present and is the Redirecting reason 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/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 26
Test purpose	Communication forwarding busy, basic rules. User A and user C are in Network A. User B is in Network B and is provided with CFB. Ensure that when user A calls user B, the call is forwarded busy to user C. In the active call state, ensure the property of speech.
Configuration	
SIP Parameter	
Message flow	

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

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

Test case number	SS_cfb_002
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	Communication forwarding busy, no notification. User A and user C are in Network A. User B is in Network B and is provided with CFB, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B, the call is forwarded busy to user C, originating user is not notified.

Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = No
SIP Parameter	
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → CFB is performed ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) Apply post test routine	
Comments	Check: No notification regarding call forwarding in Network B is received at the interconnection interface. Repeat this test in reverse direction.

Test case number	SS_cfb_003
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	Communication forwarding busy, originating user is notified. URI from the served user not received. User A and user C are in Network A. User B is in Network B and is provided with CFB Originating user receives notification that his communication has been diverted = Yes "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded busy to user C, user A is notified of call diversion and not informed of the diverted-to number and served user number.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No
SIP Parameter	181 Being Forwarded P-Asserted-Identity: <userB@NetworkB> Privacy: id History-Info: <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@networkA;cause=486*?Privacy=history*>;index=1.1

Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded (Call-IDB-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 are 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 '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. Repeat this test in reverse direction.

Test case number	SS_cfb_004
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[ETSI TS 124 604
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	Communication forwarding busy, originating user is notified. URI from the diverted-to user received. User A and user C are in Network A. User B is in Network B and is provided with CFB. Originating user receives notification that his communication has been diverted = Yes "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes. Ensure that when user A calls user B, the call is forwarded busy to user C, user A is notified of call diversion and informed of the diverted-to number.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes
SIP Parameter	181 Being Forwarded P-Asserted-Identity: <userB@NetworkB> History-Info: *<sip:userB@networkB>;index=1,* <sip: userC@networkA;cause=486>;index=1.1

Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) Apply post test routine	
Comments	Check: A 181 Being Forwarded is received at interconnection interface. Check: A History-Info header is contained in the 181 with the URI of the diverted-to user. Check: Is the "user=phone" parameter present in all History-Info header URIs? Check: Is the cause parameter in the last entry set to '486'? 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_cfb_005
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[ETSI TS 124 604
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	Communication forwarding busy, diverted-to user does not receive the URI of the served user. User A and user C are in Network C. User B is in Network B and is provided with CFB "Served user allows the presentation of his/her URI to the diverted-to user" = No. Ensure that when user A calls user B, the call is forwarded busy to user C, and user C is not informed of the forwarding number.
Configuration	Subscription options: • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE: History-Info: <sip:userB@networkB?Privacy=history& Reason=SIP%3Bcause%3D486>>;index=1, <sip: userC@networkA;cause=486>;index=1.1
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed ← INVITE(Call-ID B-C) Apply post test routine	

Comments	Check: A History-Info header is received in the INVITE and 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 set to '486'? Check: Is the "user=phone" parameter present in all History-Info header URIs? NOTE 1 – The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. NOTE 2 – The Request line may contain a 'cause' parameter indicating the redirecting reason. NOTE 3 – 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 determined user busy" by the served user. Repeat this test in reverse direction.

Test case number	SS_cfb_006
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	Communication forwarding busy, diverted-to user receives the URI of the served user. User A and user C are in Network C. User B is in Network B and is provided with CFB "Served user allows the presentation of his/her URI to the diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded busy "user determined user busy" to user C, and user C is informed of the forwarding number.
Configuration	Subscription options: • Served user allows the presentation of his/her URI to the diverted-to user = Yes
SIP Parameter	INVITE: History-Info: <sip:userB@networkB?Reason=SIP%3Bcause%3D486>;index=1, <sip: userC@networkA;cause=486>;index=1.1
Message flow SIP (Network A)	Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → CFB is performed ← INVITE(Call-ID B-C) Apply post test routine
Comments	Check:A History-Info header is received in the INVITE and 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 URIs?

	NOTE 1 – The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. NOTE 2 – The Request line may contain a 'cause' parameter indicating the redirecting reason. NOTE 3 – 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 determined user busy" by the served user. Repeat this test in reverse direction.

Test case number	SS_cfb_007
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 26 AND SE 30
Test purpose	Communication forwarding busy, full notification. User A and user C are in Network A. User B is in Network B and is provided with CFB. Originating user receives notification that his communication has been diverted = Yes "Served user allows the presentation of forwarded to URI to originating user in diversion notification"= Yes, "diverting number is released to the diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded busy to user C, and user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes • Diverting number is released to the diverted-to user = Yes
SIP Parameter	INVITE: History-Info: <sip:userB@networkB&Reason=SIP%3Bcause%3D486>;index=1, <sip: userC@networkA;cause=486>;index=1.1 181 Being Forwarded History-Info: <sip:userB@networkB&Reason=SIP%3Bcause%3D486>;index=1, <sip: userC@networkA;cause=486>;index=1.1 200 OK INVITE History-Info header: <sip:userB@networkB&Reason=SIP%3Bcause%3D486>;index=1, <sip: userC@networkA;cause=486>;index=1.1
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) ➔ CFB is performed	

← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ← ACK(Call-ID C-B) ← 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → Communication Apply post test routine	
Comments	Check:A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. Check:A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. Check:Is the cause parameter in the last entry set to '486'? Check:Is the "user=phone" parameter present in all History-Info header URIs? NOTE 1 – The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. NOTE 2 – The Request line may contain a 'cause' parameter indicating the redirecting reason. NOTE 3 – 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 determined user busy" by the served user. Repeat this test in reverse direction.

Test case number	SS_cfb_008
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 26
Test purpose	Communication forwarding busy, unsuccessful UDUB. User A and user C are in Network A. User B is in Network B and is provided with CFB. Ensure that when user A calls user B, the call is forwarded busy to user C and user C is user determined user busy.
Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed ← INVITE(Call-ID B-C)	

486 Busy Here(Call-ID C-B) → ← ACK(Call-ID B-C) ← 486 Busy Here(Call-ID A-B) ACK(Call-ID A-B) →	
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here Repeat this test in reverse direction.

Test case number	SS_cfb_009
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 26
Test purpose	Communication forwarding busy, unsuccessful NDUB. User A and user C are in Network A. User B is in Network B and is provided with CFB. Ensure that when user A calls user B, the call is forwarded busy to user C and user C is network determined user busy.
Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed ← INVITE(Call-ID B-C) 486 Busy Here(Call-ID C-B) → ← ACK(Call-ID B-C) ← 486 Busy Here(Call-ID A-B) ACK(Call-ID A-B) →	
Comments	Check: A 181 Being Forwarded is received at Network 1 originating access Check: The dialogue is terminated by receiving a 486 Busy Here Repeat this test in reverse direction.

Test case number	SS_cfb_010
Test case group	SIP-SIP/Service/CFB
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 26 AND SE 30 AND [Network A] SE 9
Test purpose	Communication forwarding busy, interaction with a non trusted network. User A and user C are in Network A. Network A is non trusted. 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, and user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) Apply post test routine	
Comments	Check: No History-Info header is received in the INVITE at the interconnection interface. Check: No History-Info header is received in the 181 Being Forwarded at the interconnection interface (if sent). Repeat this test in reverse direction.

Test case number	SS_cfb_011
Test case group	SIP-SIP/Service/CFB
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55A
Test purpose	SIP-I support. CFB performed in Network B, Notification subscription options is set to presentation not allowed. User A and user C are in Network A. 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 = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on busy user to user C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = no

<table>
<tr><td>SIP Parameter</td><td>183 Session Progress
    Content-Type: multipart/mixed;boundary=[any boundary name]

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

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

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

Test case number	SS_cfb_012
Test case group	SIP-SIP/Service/CFB
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND 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. User A and user C are in Network A. 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 = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on busy user to user C, and user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed without redirection number Redirecting reason User Busy Generic notification call is diverting --[any boundary name]--

Message flow	
SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: 183 Session Progress is received at the interconnection interface. Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND 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. User A and user C are in Network A. 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 = yes, with diverted-to user number. Ensure that when user A calls user B, the call is forwarded on busy user to user C, and 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

<table>
<tr><td>SIP Parameter</td><td>183 Session Progress
    Content-Type: multipart/mixed;boundary=[any boundary name]

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

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

    --[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →
CFB is performed
← INVITE(Call-ID B-C, IAM)
← 183 Session Progress (Call-ID B-A, ACM)
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A.
Check: 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 'User Busy'?
Repeat this test in reverse direction.</td></tr>
</table>

<table>
<tr><td>Test case number</td><td>SS_cfb_014</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CFB</td></tr>
<tr><td>Reference</td><td>6.7/[ITU-T Q.1912.5]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 17 AND SE 47 AND SE 53A</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. CFB performed in Network B, Restriction of the Redirection number
User A and user C are in Network A. 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.</td></tr>
<tr><td>Configuration</td><td>Subscription options:
• Connected user subscribed to COLR, Permanent = yes</td></tr>
<tr><td>SIP Parameter</td><td>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]--</td></tr>
<tr><td colspan="2">Message flow
<table>
<tr><td>SIP (Network A)</td><td></td><td>Interconnection Interface</td><td></td><td>SIP (Network B)</td></tr>
<tr><td></td><td></td><td>INVITE(Call-ID A-B)</td><td>→</td><td></td></tr>
<tr><td></td><td></td><td>CFB is performed</td><td></td><td></td></tr>
<tr><td></td><td>←</td><td>INVITE(Call-ID B-C, IAM)</td><td></td><td></td></tr>
<tr><td></td><td></td><td>180 Ringing (Call-ID C-B)</td><td>→</td><td></td></tr>
<tr><td></td><td>←</td><td>180 Ringing (Call-ID B-A, ACM)</td><td></td><td></td></tr>
<tr><td></td><td></td><td>200 OK INVITE (Call-ID C-B)</td><td>→</td><td></td></tr>
<tr><td></td><td>←</td><td>ACK (Call-ID B-C)</td><td></td><td></td></tr>
<tr><td></td><td>←</td><td>200 OK INVITE (Call-ID B-A, ANM)</td><td></td><td></td></tr>
<tr><td></td><td></td><td>ACK (Call-ID A-B)</td><td>→</td><td></td></tr>
<tr><td></td><td></td><td>Apply post test routine</td><td></td><td></td></tr>
</table></td></tr>
<tr><td>Comments</td><td>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.</td></tr>
</table>

Test case number	SS_cfb_015
Test case group	SIP-SIP/Service/CFB
Reference	6.7/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53A
Test purpose	SIP-I support. CFB performed in Network B, No restriction of the Redirection number. User A and user C are in Network A. User B is in the PSTN/PLMN part of Network B and is provided with CFB. Diverted-to user is not subscribed to the COLR service. Ensure that when user A calls user B, the call is forwarded on busy user to user C, if a Redirection number restriction parameter is present it is set to 'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	Subscription options: • Connected user subscribed to COLR = no
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM Redirection number restriction Presentation allowed or Redirection number restriction not present --[any boundary name]--

Message flow			
SIP (Network A)	Interconnection Interface		SIP (Network B)
	INVITE(Call-ID A-B)	→	
	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)	→	
	Apply post test routine		

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

Test case number	SS_cfb_016
Test case group	SIP-SIP/Service/CFB
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55A
Test purpose	SIP-I support. CFB performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed' User A and user C are in Network A. 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, and user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC – SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason

	User Busy --[any boundary name]--
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFB is performed ← INVITE(Call-ID B-C, IAM) Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and is the Address presentation restricted indicator set to 'presentation allowed'? Check: Is the Original called number present and is the Address presentation restricted indicator set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is Redirection information present and is the Redirecting reason set to 'User Busy'? Repeat this test in reverse direction.

Test case number	SS_cfb_017
Test case group	SIP-SIP/Service/CFB
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55A
Test purpose	SIP-I support. CFB performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted' User A and user C are in Network A. 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 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 restricted Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason User Busy --[any boundary name]--
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) ➔ CFB is performed 🡐 INVITE(Call-ID B-C, IAM) Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and is the Address presentation restricted indicator set to 'presentation restricted'? Check: Is the Original called number present and is the Address presentation restricted indicator set to 'presentation restricted'? Check: Is the Redirection number present? Check: Is Redirection information present and is the Redirecting reason 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/[ETSI TS 124 604]

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

Test case number	SS_cfnr_002
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 27 AND SE 30
Test purpose	Communication forwarding no reply, no notification. User A and user C are in Network A. 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, and 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/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 27 AND SE 30
Test purpose	Communication forwarding no reply, originating user is notified. URI from the served user not received. User A and user C are in Network A. 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, and 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> Privacy: id History-Info: <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@networkA;cause=408?Privacy=history>;index=1.1
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing(Call-ID B-A) CFB is performed	

	← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) Apply post test routine
Comments	Check: A 181 Being Forwarded and a History-Info header are received at the interconnection interface; in both entries in the History-Info header a Privacy header is escaped value 'history' Check: Is the cause parameter in the last entry is set to '408'? 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_cfnr_004
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 27 AND SE 30
Test purpose	Communication forwarding no reply, originating user is notified. URI from the diverted-to user received. User A and user C are in Network A. 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, and 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> History-Info: <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=408>;index=1.1

Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing(Call-ID B-A) CFB is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) Apply post test routine	
Comments	Check: A 181 Being Forwarded is received at the interconnection interface. Check: A History-Info header is contained in the 181 with the URI of the diverted-to user. Check: Is the cause parameter in the last entry 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.

Test case number	SS_cfnr_005
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 27 AND SE 30
Test purpose	Communication forwarding no reply, diverted-to user does not receive the URI of the served user. User A and user C are in Network A. User B is in Network B and is provided with "Served user allows the presentation of his/her URI to the diverted-to user" = No. Ensure that when user A calls user B, the call is forwarded no reply to user C, and user C is not informed of the forwarding number.
Configuration	Subscription options: • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE History-Info: <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@network1;cause=408>;index=1.1

Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing(Call-ID B-A) CFB is performed ← INVITE(Call-ID B-C) Apply post test routine	
Comments	Check: A History-Info header is received in the INVITE and 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 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 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_cfnr_006
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 27 AND SE 30
Test purpose	Communication forwarding no reply, diverted-to user receives the URI of the diverted-to user. User A and user C are in Network A. User B is in Network B and is provided with "Served user allows the presentation of his/her URI to the diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded no reply to user C, user C is informed of the forwarding number.
Configuration	Subscription options: • Served user allows the presentation of his/her URI to the diverted-to user = Yes
SIP Parameter	INVITE History-Info: <sip:userB@networkB>;index=1, <sip: userC@network1;cause=408>;index=1.1
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing(Call-ID B-A) CFB is performed ← INVITE(Call-ID B-C) Apply post test routine	

Comments	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 1 – The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. NOTE 2 – The Request line may contain a 'cause' parameter indicating the redirecting reason. Repeat this test in reverse direction.

Test case number	SS_cfnr_007
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 27 AND SE 30
Test purpose	Communication forwarding no reply, full notification. User A and user C are in Network A. User B is in Network B and is provided with CFNR. Originating user receives notification that his communication has been diverted = Yes, "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded no reply to user C, and user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes • Diverting number is released to the diverted-to user = Yes
SIP Parameter	INVITE: History-Info: <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=486>;index=1.1 181 Being Forwarded P-Asserted-Identity: <userB@NetworkB> History-Info: <sip:userB@network>;index=1, <sip: userC@networkA;cause=408>;index=1.1 200 OK INVITE

	History-Info: <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=408>;index=1.1
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing(Call-ID B-A) CFB is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) →	
	← ACK(Call-ID C-B) ← 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → Apply post test routine
Comments	Check: A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user. Check: A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user. Check: Is the cause parameter 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 1 – The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. NOTE 2 – The Request line may contain a 'cause' parameter indicating the redirecting reason. Repeat this test in reverse direction.

Test case number	SS_cfnr_008
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 27
Test purpose	Communication forwarding no reply, unsuccessful UDUB. User A and user C are in Network A. User B is in Network B and is provided with CFNR. Ensure that when user A calls user B, the call is forwarded no reply to user C and user C is user determined user busy.
Configuration	
SIP Parameter	

Message flow			
SIP (Network A)	Interconnection Interface		SIP (Network B)
	INVITE(Call-ID A-B)	→	
←	180 Ringing(Call-ID B-A)		
	CFB is performed		
←	INVITE(Call-ID B-C)		
	486 Busy Here(Call-ID C-B)	→	
←	ACK(Call-ID B-C)		
←	486 Busy Here(Call-ID A-B)		
	ACK(Call-ID A-B)	→	
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here Repeat this test in reverse direction.		

Test case number	SS_cfnr_009
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[ETSI TS 124 604
SELECTION EXPRESSION	SE 27
Test purpose	Communication forwarding no reply, unsuccessful NDUB. User A and user C are in Network A. User B is in Network B and is provided with CFNR. Ensure that when user A calls user B, the call is forwarded no reply to user C and user C is network determined user busy.
Configuration	
SIP Parameter	

Message flow			
SIP (Network A)	Interconnection Interface		SIP (Network B)
	INVITE(Call-ID A-B)	→	
←	180 Ringing(Call-ID B-A)		
	CFB is performed		
←	INVITE(Call-ID B-C)		
	486 Busy Here(Call-ID C-B)	→	
←	ACK(Call-ID B-C)		
←	486 Busy Here(Call-ID A-B)		
	ACK(Call-ID A-B)	→	
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here Repeat this test in reverse direction.		

Test case number	SS_cfnr_010
Test case group	SIP-SIP/Service/CFNR
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 27 AND SE 30 AND [Network A] is SE 9

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

Test case number	SS_cfnr_011
Test case group	SIP-SIP/Service/CFNR
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55B
Test purpose	SIP-I support. CFNR performed in Network B, Notification subscription options is set to presentation ‘not allowed’. User A and user C are in Network A. 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" = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on no reply to user C, user A is not notified about call diversion.

	The notification information is present in the encapsulated CPG contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = no
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Event indicator Alerting or Progress Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation not allowed Redirecting reason No reply Generic notification call is diverting --[any boundary name]--
Message flow SIP (Network A)	Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A, ACM) CFNR is performed ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, CPG) Apply post test routine
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is a 183 Session Progress received at the interconnection interface? Check: Is an CPG encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator set to 'presentation not allowed'? Check: Is the Redirecting reason set to 'No reply'? Repeat this test in reverse direction.

Test case number	SS_cfnr_012
Test case group	SIP-SIP/Service/CFNR
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND 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 User A and user C are in Network A. 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" = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on no reply to user C, and user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated CPG contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Event indicator Alerting or Progress Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed without redirection number Redirecting reason No reply Generic notification call is diverting --[any boundary name]--

Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A, ACM) CFNR is performed ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: 183 Session Progress is received at the interconnection interface. Check: Is an CPG encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND 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. User A and user C are in Network A. 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" = yes, with diverted-to user number. Ensure that when user A calls user B, the call is forwarded on no reply to user C, and 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

<table>
<tr><td></td><td>Alerting or Progress
Redirection number
Address signal (Diverted-to user)
Call diversion information
Notification subscription options
presentation allowed with redirection number
Redirecting reason
No reply
Generic notification
call is diverting

--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →
← 180 Ringing (Call-ID B-A, ACM)
CFNR is performed
← INVITE(Call-ID B-C, IAM)
← 183 Session Progress (Call-ID B-A, ACM)
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A.
Check: 183 Session Progress is received at the interconnection interface.
Check: Is 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 allowed with redirection number'?
Check: Is the Redirecting reason set to 'No reply'?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_cfnr_014
Test case group	SIP-SIP/Service/CFNR
Reference	6.7/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53B
Test purpose	SIP-I support. CFNR performed in Network B, Restriction of the Redirection number. User A and user C are in Network A. User B is in the PSTN/PLMN part of Network B and is provided with CFNR. Diverted-to user is subscribed to the COLR service in Permanent mode. Ensure that when user A calls user B, the call is forwarded on no reply to user C; a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.

Configuration	Subscription options: • Connected user subscribed to COLR, Permanent = yes
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM Redirection number restriction Presentation restricted --[any boundary name]--
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → ← 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 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 53B

<table>
<tr><td>Test purpose</td><td>SIP-I support. CFNR performed in Network B. No restriction of the Redirection number
User A and user C are in Network A. 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.</td></tr>
<tr><td>Configuration</td><td>Subscription options:
• Connected user subscribed to COLR = no</td></tr>
<tr><td>SIP Parameter</td><td>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]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →
← 180 Ringing (Call-ID B-A)
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</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A.
Check: Is a 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.</td></tr>
</table>

<table>
<tr><td>Test case number</td><td>SS_cfnr_016</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CFNR</td></tr>
<tr><td>Reference</td><td>7.1/[ITU-T Q.1912.5]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network B] SE 17 AND SE 47 AND SE 55B</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. CFNR performed in Network B. Notification of diverted-to user Redirecting number 'presentation allowed'.
User A and user C are in Network A. 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, and user C is notified of call diversion and informed of the diverting number.
The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC – SIP-I interworking is applicable in Network B.</td></tr>
<tr><td>Configuration</td><td>Subscription options:
• Served user releases his/her number to diverted-to user = Release diverting number information</td></tr>
<tr><td>SIP Parameter</td><td>INVITE
    Content-Type: multipart/mixed;boundary=[any boundary name]

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

      IAM
        Redirecting number
          Address presentation restricted indicator
            presentation allowed
          Address signal (Diverting user)
        Original called number
          Address presentation restricted indicator
            presentation allowed
          Address signal
        Redirection information
          Original Redirection Reason
            unknown
          Redirecting indicator
          Redirection counter
          Redirecting reason
            No reply

    --[any boundary name]--</td></tr>
</table>

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 is the Address presentation restricted indicator set to 'presentation allowed'? Check: Is the Original called number present and is the Address presentation restricted indicator set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is Redirection information present and is the Redirecting reason set to 'No reply'? Repeat this test in reverse direction.

Test case number	SS_cfnr_017
Test case group	SIP-SIP/Service/CFNR
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55B
Test purpose	SIP-I support. CFNR performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted' User A and user C are in Network A. 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, and user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if ISUP/BICC – SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Do not release diverting number information
SIP Parameter	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

<table>
<tr><td></td><td>Address presentation restricted indicator
presentation restricted
Address signal (Diverting user)
Original called number
Address presentation restricted indicator
presentation restricted
Address signal
Redirection information
Original Redirection Reason
unknown
Redirecting indicator
Redirection counter
Redirecting reason
No reply

--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →
← 180 Ringing (Call-ID B-A, ACM)
CFNR is performed
← INVITE(Call-ID B-C, IAM)
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A.
Check: Is an INVITE request received at the interconnection interface?
Check: Is an IAM encapsulated in the INVITE?
Check: Is the Redirecting number present and is the Address presentation restricted indicator set to 'presentation restricted'?
Check: Is the Original called number present and is the Address presentation restricted indicator set to 'presentation restricted'?
Check: Is the Redirection number present?
Check: Is Redirection information present and is the Redirecting reason set to 'No reply'?
Repeat this test in reverse direction.</td></tr>
</table>

7.1.5.6.4 Communication Forwarding Not Logged in (CFNL)

Test case number	SS_cfnl_001
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 28
Test purpose	Communication forwarding not logged in, basic rules. User A and user C are in Network A. User B is in Network B and is provided with CFNL. Ensure that when user A calls user B, the call is forwarded not logged in to user C. In the active call state, ensure the property of speech.

Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFNL is performed ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ← ACK(Call-ID B-C) ← 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → Communication Apply post test routine	
Comments	Check: The CDIV not logged in is successful Check: In the active call state, ensure the property of speech Check: Is the P-Asserted-Identity present in the INVITE sent from Network B to Network A set to the identity of the originating user? Repeat this test in reverse direction.

Test case number	SS_cfnl_002
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	Communication forwarding not logged in, no notification. User A and user C are in Network A. User B is in Network B and is provided with CFNL, subscription option: "Originating user receives notification that his communication has been diverted" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, originating user is not notified.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = No
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFNL is performed ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) Apply post test routine	

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

Test case number	SS_cfnl_003
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	Communication forwarding not logged in, originating user is notified. URI of the diverted-to user not received. User A and user C are in Network A. User B is in Network B and is provided with CFNL. Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and not informed of the diverted-to number and the served user number.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No
SIP Parameter	181 Being Forwarded P-Asserted-Identity: <userB@NetworkB> Privacy: id History-Info: *<sip:userB@networkB?Privacy=history>;index=1,* *<sip: userC@networkA;cause=404?Privacy=history>;index=1.1*
Message flow SIP (Network A) | Interconnection Interface | SIP (Network B) INVITE(Call-ID A-B) → CFNL is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) Apply post test routine	
Comments	Check: A 181 Being Forwarded and a History-Info header are 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/[ETSI TS 124 604]
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. User A and user C are in Network A. 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, and 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> History-Info: *<sip:userB@networkB>;index=1,* <sip: userC@networkA;cause=404>;index=1.1
Message flow SIP (Network A)	Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → CFNL is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) Apply post test routine
Comments	Check: A 181 Being Forwarded is received at interconnection interface. Check: A History-Info header is contained in the 181 with the URI of the served user and the URI of the diverted-to user. Check: Is the cause parameter in the last entry 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_005
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	Communication forwarding not logged in, diverted-to user does not receive the URI of the diverted-to user. User A and user C are in Network A. User B is in Network B and is provided with CFNL "Served user allows the presentation of his/her URI to diverted-to user" = No. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user C is not informed of the forwarding number.
Configuration	Subscription options: • Served user allows the presentation of his/her URI to diverted-to user = No
SIP Parameter	INVITE History-Info: <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@network1;cause=404>;index=1.1
Message flow SIP (Network A)	Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → CFNL is performed ← INVITE(Call-ID B-C) Apply post test routine
Comments	Check: A History-Info header is received in the INVITE and 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 set to '404'? Check: Is the "user=phone" parameter present in all History-Info header URIs? NOTE 1 – The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. NOTE 2 – The Request line may contain a 'cause' parameter indicating the redirecting reason. Repeat this test in reverse direction.

Test case number	SS_cfnl_006
Test case group	SIP-SIP/Service/CFNL

Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	Communication forwarding not logged in, diverted-to user receives the URI of the served user. User A and user C are in Network A. User B is in Network B and is provided with CFNL "Served user allows the presentation of his/her URI to diverted-to user" = Yes. Ensure that when user A calls user B, the call is forwarded not logged in to user C, user C is informed of the forwarding number.
Configuration	Subscription options: • Served user allows the presentation of his/her URI to diverted-to user = Yes
SIP Parameter	INVITE History-Info: *<sip:userB@networkB>;index=1,* <sip: userC@networkA;cause=404>;index=1.1
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFNL is performed ← INVITE(Call-ID B-C) Apply post test routine	
Comments	Check: A History-Info header is received in the INVITE and contains the URI of user B (served user) at the interconnection interface. Check: Is the cause parameter in the last entry set to '404'? Check: Is the "user=phone" parameter present in all History-Info header URIs? NOTE 1 – The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. NOTE 2 – The Request line may contain a 'cause' parameter indicating the redirecting reason. Repeat this test in reverse direction.

Test case number	SS_cfnl_007
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 28 AND SE 30
Test purpose	Communication forwarding not logged in, full notification. User A and user C are in Network A. User B is in Network B and is provided with CFNL. Originating user receives notification that his communication has been diverted = Yes, "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes.)

<table>
<tr><td></td><td>Ensure that when user A calls user B, the call is forwarded not logged in to user C, and user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number.</td></tr>
<tr><td>Configuration</td><td>Subscription options:
• Originating user receives notification that his communication has been diverted = Yes
• Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes
• diverting number is released to the diverted-to user = Yes</td></tr>
<tr><td>SIP Parameter</td><td>INVITE:
History-Info:
<sip:userB@networkB&Reason=SIP;cause=404>;index=1,
<sip: userC@networkA;cause=404>;index=1.1

181 Being Forwarded
P-Asserted-Identity: <userB@NetworkB>
History-Info:
<sip:userB@network>;index=1,
<sip: userC@networkA;cause=404>;index=1.1

200 OK INVITE
History-Info:
<sip:userB@networkB>;index=1,
<sip: userC@networkA;cause=404>;index=1.1</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →
CFNL is performed
← INVITE(Call-ID B-C)
← 181 Being Forwarded(Call-ID B-A
180 Ringing(Call-ID C-B) →
← 180 Ringing(Call-ID B-A)
200 OK INVITE(Call-ID C-B) →
← ACK(Call-ID C-B)
← 200 OK INVITE(Call-ID B-A)
ACK(Call-ID A-B) →
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Check: A History-Info header is received in the INVITE at the interconnection interface sent to user C containing the URI identifying the served user.
Check: A History-Info header is received in the 181 Being Forwarded at the interconnection interface sent to user A containing the URI identifying the diverted-to user.
Check: Is the cause parameter in the last entry 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?</td></tr>
</table>

<table>
<tr><td></td><td>NOTE 1 – The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.
NOTE 2 – The Request line may contain a ‘cause’ parameter indicating the redirecting reason.
Repeat this test in reverse direction.</td></tr>
</table>

<table>
<tr><td>Test case number</td><td>SS_cfnl_008</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CFNL</td></tr>
<tr><td>Reference</td><td>4.5.2.6/[ETSI TS 124 604]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>SE 28</td></tr>
<tr><td>Test purpose</td><td>Communication forwarding not logged in, unsuccessful UDUB.
User A and user C are in Network A. 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.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td></td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(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) →</td></tr>
<tr><td>Comments</td><td>Check: The dialogue is terminated by receiving a 486 Busy Here.
Repeat this test in reverse direction.</td></tr>
</table>

<table>
<tr><td>Test case number</td><td>SS_cfnl_009</td></tr>
<tr><td>Test case group</td><td>4.5.2.6/[ETSI TS 124 604]</td></tr>
<tr><td>Reference</td><td>ES 183 004</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>SE 28</td></tr>
<tr><td>Test purpose</td><td>Communication forwarding not logged in, unsuccessful NDUB.
User A and user C are in Network A. 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.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td></td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →</td></tr>
</table>

CFNL is performed 486 Busy Here(Call-ID C-B) → ← ACK(Call-ID B-C) ← 486 Busy Here(Call-ID A-B) ACK(Call-ID A-B) →	
Comments	Check: The dialogue is terminated by receiving a 486 Busy Here Repeat this test in reverse direction.

Test case number	SS_cfnl_010
Test case group	SIP-SIP/Service/CFNL
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 28 AND SE 30 AND [Network A] SE 9
Test purpose	Communication forwarding not logged in, interaction with a non trusted network. User A and user C are in Network A. Network A is non trusted. 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, and 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND 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. User A and user C are in Network A. 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 = yes, without diverted-to user number. Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C; user A is not notified about call diversion. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = no
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation not allowed Redirecting reason Mobile subscriber not reachable Generic notification call is diverting --[any boundary name]--

Message flow	
SIP (Network A) · Interconnection Interface · SIP (Network B) INVITE(Call-ID A-B) → CFNL is performed ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is a 183 Session Progress received at the interconnection interface? Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator set to 'presentation not allowed'? Check: Is the Redirecting reason set to 'Mobile subscriber not reachable'? Repeat this test in reverse direction.

Test case number	SS_cfnl_012
Test case group	SIP-SIP/Service/CFNL
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND 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. User A and user C are in Network A. 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 = 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

<table>
<tr><td>SIP Parameter</td><td>183 Session Progress
Content-Type: multipart/mixed;boundary=[any boundary name]

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

ACM
Backward call indicator
Called party's status indicator
no indication
Redirection number
Address signal (Diverted-to user)
Call diversion information
Notification subscription options
presentation allowed without redirection number
Redirecting reason
Mobile subscriber not reachable
Generic notification
call is diverting

--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
INVITE(Call-ID A-B) →
CFNL is performed
← INVITE(Call-ID B-C, IAM)
← 183 Session Progress (Call-ID B-A, ACM)
Apply post test routine</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A.
Check: 183 Session Progress is received at the interconnection interface.
Check: Is an ACM encapsulated in the 183?
Check: Is the Called party's status indicator set to 'no indication'?
Check: Is the Redirection number present?
Check: Is Notification subscription options indicator set to 'presentation allowed without redirection number'?
Check: Is the Redirecting reason set to 'Mobile subscriber not reachable'?
Repeat this test in reverse direction.</td></tr>
</table>

Test case number	SS_cfnl_013
Test case group	SIP-SIP/Service/CFNL
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND 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. User A and user C are in Network A. 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 = 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, and user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number.
SIP Parameter	183 Session Progress Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator no indication Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed with redirection number Redirecting reason Mobile subscriber not reachable Generic notification call is diverting --[any boundary name]--

Message flow	
SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CFNL is performed ← INVITE(Call-ID B-C, IAM) ← 183 Session Progress (Call-ID B-A, ACM) Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A Check: 183 Session Progress is received at the interconnection interface. Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 55C
Test purpose	SIP-I support. 'Mobile subscriber not reachable' performed in Network B, Restriction of the Redirection number. User A and user C are in Network A. 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 'Mobile subscriber not reachable to user C, and a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC-SIP-I interworking is applicable in Network A.
Configuration	Subscription options: • Connected user subscribed to COLR, Permanent = yes
SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM Redirection number restriction Presentation restricted --[any boundary name]--

Message flow	
SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → 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) → 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 55C
Test purpose	SIP-I support. 'Mobile subscriber not reachable' performed in Network B, No restriction of the Redirection number. User A and user C are in Network A. 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, and 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

<table>
<tr><td>SIP Parameter</td><td>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]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
INVITE(Call-ID A-B) →
CFNL is performed
← INVITE(Call-ID B-C) , IAM
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</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B.
Network B performs the diversion to a user in Network A.
Check: Is a 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.</td></tr>
</table>

Test case number	SS_cfnl_016
Test case group	SIP-SIP/Service/CFNL
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND 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'. User A and user C are in Network A. 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, and user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC – SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information.
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason Mobile subscriber not reachable --[any boundary name]--

Message flow		
SIP (Network A)	Interconnection Interface	SIP (Network B)
	INVITE(Call-ID A-B) →	
	CFNL is performed	
	← INVITE(Call-ID B-C, IAM)	
	Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and is the Address presentation restricted indicator set to 'presentation allowed'? Check: Is the Original called number present and is the Address presentation restricted indicator set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is Redirection information present and is the Redirecting reason 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND 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' User A and user C are in Network A. 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, and user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if ISUP/BICC – SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Do not release diverting number information
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number

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

7.1.5.6.5 Communication deflection

Test case number	SS_cd_001
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 29
Test purpose	Communication deflection during alerting, basic rules. User A and user C are in Network A. User B is in Network B and is provided with CDa. Ensure that when user A calls user B, the call is deflected during alerting to user C. In the active call state, ensure the property of speech.
Configuration	

SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CDa is performed ← 180 Ringing(Call-ID B-A) ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ← ACK(Call-ID B-C) ← 200 OK INVITE(Call-ID B-A) ACK(Call-ID A-B) → Communication Apply post test routine	
Comments	Check: CDa is successful. Check: In the active call state, ensure the property of speech. Check: Is the P-Asserted-Identity present set to the identity of the originating user? Repeat this test in reverse direction.

Test case number	SS_cd_002
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 29
Test purpose	Communication deflection immediate, basic rules. User A and user C are located in Network A. User B is located in network B and is provided with CDi. Ensure that when user A calls user B, which immediately deflects the communication towards user C (i.e., before alerting starts), the call is forwarded to user C. In the active call state, ensure the property of speech.
Configuration	
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → CDi is performed ← INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) → ← 180 Ringing(Call-ID B-A) 200 OK INVITE(Call-ID C-B) → ← ACK(Call-ID B-C) ← 200 OK INVITE(Call-ID B-A)	

ACK(Call-ID A-B) ➔ Communication Apply post test routine	
Comments	Check: CDi is successful. Check: In the active call state, ensure the property of speech. Check: Is the P-Asserted-Identity present in the INVITE sent from Network B to Network A set to the identity of the originating user? Repeat this test in reverse direction.

Test case number	SS_cd_003
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 29 AND SE 30
Test purpose	Communication Deflection immediate response, no notification. User A and user C are in Network A. User B is in Network B and is provided with CDi, subscription option: Originating user receives notification that his communication has been diverted = No. Ensure that when user A calls user B. which deflects immediately the communication towards user C (i.e., before alerting starts), the call is forwarded to user C. Ensure that User A does not receive a 181 Call Is Being Forwarded message.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = No
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) ➔ CDi is performed 🡐 INVITE(Call-ID B-C) 180 Ringing(Call-ID C-B) ➔ 🡐 180 Ringing(Call-ID B-A) Apply post test routine	
Comments	Check: No notification regarding call forwarding in Network B is received at the interconnection interface. Check: Is the cause parameter in the last entry is set to '480'? Repeat this test in reverse direction.

Test case number	SS_cd_004
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[ETSI TS 124 604]
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. User A and user C are located in Network A. User B is located in Network B and is provided with CDi Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No.) Ensure that when user A calls user B, which deflects immediately the communication towards user C (i.e., before alerting starts), the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message, and user A is notified of call diversion and not informed of the diverted-to number and served user number.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Originating user receives notification that his communication has been diverted = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No
SIP Parameter	181 Being Forwarded History-Info: P-Asserted-Identity: <userB@NetworkB> Privacy: id <sip:userB@networkB?Privacy=history>;index=1, <sip: userC@networkA;cause=480?Privacy=history>;index=1.1
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) ➔ CDi is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) Apply post test routine	
Comments	Check: A 181 Being Forwarded and a History-Info header is received at the interconnection interface in both entries in the History-Info header a Privacy header is escaped value 'history'. Check: Is the cause parameter in the last entry is set to '480'? 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_005
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 29 AND SE 30
Test purpose	Communication Deflection immediate response, originating user is notified. URI from the diverted-to user received. User A and user C are in Network A. User B is in Network B and is provided with CDi. Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes. Ensure that when user A calls user B, which deflects immediately the communication towards user C (i.e., before alerting starts), the call is forwarded to user C. Ensure that User A receives a 181 Call Is Being Forwarded message, user A is notified of call diversion and informed of the diverted-to number.
Configuration	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of diverted to URI to originating user in diversion notification = Yes
SIP Parameter	181 Being Forwarded P-Asserted-Identity: <userB@NetworkB> History-Info: <sip:userB@networkB>;index=1, <sip: userC@networkA;cause=480>;index=1.1
Message flow SIP (Network A) | Interconnection Interface | SIP (Network B) INVITE(Call-ID A-B) → CDi is performed ← INVITE(Call-ID B-C) ← 181 Being Forwarded(Call-ID B-A) Apply post test routine	
Comments	Check: A 181 Being Forwarded is received at the interconnection interface. Check: A History-Info header is contained in the 181 with the URI of the diverted-to user Check: Is the cause parameter in the last entry 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/[ETSI TS 124 604]
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. User A and user C are in Network A. 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
Message flow SIP (Network A)	Interconnection Interface / SIP (Network B) INVITE(Call-ID A-B) ➔ CDi is performed 🡐 INVITE(Call-ID B-C) Apply post test routine
Comments	Check: A History-Info header is received in the INVITE contains the URI of user B (served user) at the interconnection interface and a Privacy header is escaped set to 'history'. Check: Is the cause parameter in the last entry is set to '480'? Check: Is the “user=phone” parameter present in all History-Info header URIs? NOTE 1 – The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. NOTE 2 – 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/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 29 AND SE 30
Test purpose	Communication Deflection immediate response, diverted-to user receives the URI of the served user. User A and user C are in Network A. UserUser A and user C are in Network A. User B is in Network B and is provided with CDi "Served user allows the presentation of his/her URI to diverted-to user" = Yes. Ensure that when user A calls user B, which deflects immediately the communication towards user C (i.e., before alerting starts), the call is forwarded to user C, user C is informed of the forwarding number.
Configuration	Subscription options: • Served user allows the presentation of his/her URI to diverted-to user = Yes
SIP Parameter	INVITE History-Info: <sip:userB@networkB?Reason=SIP%3Bcause%3D302>;index=1, <sip: userC@networkA;cause=480>;index=1.1
Message flow	SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) ➔ CDi is performed 🡐 INVITE(Call-ID B-C) Apply post test routine
Comments	Check: A History-Info header is received in the INVITE and 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 1 – The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header. NOTE 2 – The Request line may contain a ‘cause’ parameter indicating the redirecting reason. Repeat this test in reverse direction.

Test case number	SS_cd_008
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 29
Test purpose	Communication Deflection immediate response, unsuccessful UDUB. User A and user C are in Network A. UserUser A and user C are in Network A. User B is in Network B and is provided with CDi. Ensure that when user A calls user B, the call is deflected immediately 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)	→	
		CDi is performed		
	←	INVITE(Call-ID B-C)		
		486 Busy Here(Call-ID C-B)	→	
	←	ACK(Call-ID B-C)		
	←	486 Busy Here(Call-ID B-A)		
		ACK(Call-ID A-B)	→	
		Apply post test routine		

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

Test case number	SS_cd_009
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 29
Test purpose	Communication Deflection immediate response, unsuccessful NDUB. User A and user C are in Network A. User B is in Network B. Ensure that when user A calls user B, the call is deflected immediately to user C and user C is network determined user busy.
Configuration	
SIP Parameter	

Message flow

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

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

Test case number	SS_cd_010
Test case group	SIP-SIP/Service/CD
Reference	4.5.2.6/[ETSI TS 124 604]
SELECTION EXPRESSION	SE 29 AND SE 30 AND [Network A] SE 9
Test purpose	Communication Deflection immediate response, interaction with a non trusted network. User A and user C are in Network A. Network A is non trusted. 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 as immediate response to user C, user A is notified of call diversion and not informed of the diverted-to number and user C is not informed of the forwarding number.
Configuration	
SIP Parameter	Subscription options: • Originating user receives notification that his communication has been diverted = Yes • Served user allows the presentation of forwarded to URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to originating user in diversion notification = No • Served user allows the presentation of his/her URI to the diverted-to user = No
SIP Parameter	INVITE: no History-Info header 181 Being Forwarded no History-Info header
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) ➔ CDi is performed 🡐 INVITE(Call-ID B-C) 🡐 181 Being Forwarded(Call-ID B-A) Apply post test routine	
Comments	Check: No History-Info header is received in the INVITE at the interconnection interface. Check: No History-Info header is received in the 181 Being Forwarded at the interconnection interface. Repeat this test in reverse direction.

Test case number	SS_cd_011
Test case group	SIP-SIP/Service/CD
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55D
Test purpose	SIP-I support. CD performed in Network B, Notification subscription options is set to presentation not allowed. User A and user C are in Network A. 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 = yes, without diverted-to user number. Ensure that when user A calls user B, the call is deflected to user C, user A is not notified about call diversion. The notification information is present in the encapsulated ACM or CPG contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = no
SIP Parameter	183/180 Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM/CPG Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation not allowed Redirecting reason Deflection immediate or Deflection during alerting Generic notification call is diverting --[any boundary name]--

Message flow

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

Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: Is a 183 Session Progress received at the interconnection interface? Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator set to 'presentation not allowed'? Check: Is the Redirecting reason set to 'Deflection immediate' or 'Deflection during alerting'? Repeat this test in reverse direction.

Test case number	SS_cd_012
Test case group	SIP-SIP/Service/CD
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND 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. User A and user C are in Network A. 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 = yes, without diverted-to user number. Ensure that when user A calls user B, the call is deflected to user C; user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number
SIP Parameter	183/180 Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM/CPG Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed without redirection number Redirecting reason Deflection immediate or Deflection during alerting Generic notification call is diverting

	--[any boundary name]--
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A) in case CDa CD is performed ← INVITE(Call-ID B-C, IAM) ← 183/180 (Call-ID B-A, ACM/CPG) Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A. Check: 183 Session Progress is received at the interconnection interface. Check: Is an ACM encapsulated in the 183? Check: Is the Called party's status indicator set to 'no indication'? Check: Is the Redirection number present? Check: Is Notification subscription options indicator 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND 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. User A and user C are in Network A. 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 = yes, with diverted-to user number. Ensure that when user A calls user B, the call is deflected to user C; user A is notified of call diversion and informed of the diverted-to number. The notification information is present in the encapsulated ACM contained in the Redirection number and Call diversion information if SIP-I – ISUP/BICC interworking is applicable in Network B.
Configuration	Subscription options: • Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number
SIP Parameter	183/180 Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM/CPG

	Redirection number Address signal (*Diverted-to user*) Call diversion information Notification subscription options presentation allowed with redirection number Redirecting reason Deflection immediate or Deflection during alerting Generic notification call is diverting --[any boundary name]--
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A, 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: 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 '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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 55D
Test purpose	SIP-I support. CD performed in Network B, Restriction of the Redirection number User A and user C are in Network A. User B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Diverted-to user is subscribed to the COLR service in Permanent mode. Ensure that when user A calls user B, the call is deflected to user C, a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.
Configuration	Subscription options: • Connected user subscribed to COLR, Permanent = yes

SIP Parameter	200 OK Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM Redirection number restriction Presentation restricted --[any boundary name]--
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A) 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) 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_cd_015
Test case group	SIP-SIP/Service/CD
Reference	6.7/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 55D
Test purpose	SIP-I support. CD performed in Network B, No restriction of the Redirection number. User A and user C are in Network A. 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.

<table>
<tr><td>Configuration</td><td>Subscription options:
• Connected user subscribed to COLR = no</td></tr>
<tr><td>SIP Parameter</td><td>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]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) | Interconnection Interface | SIP (Network B)
INVITE(Call-ID A-B) →
← 180 Ringing (Call-ID B-A) in case CDa
CD is performed
← INVITE(Call-ID B-C, IAM)
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</td></tr>
<tr><td>Comments</td><td>Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A.
Check: Is a 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.</td></tr>
</table>

Test case number	SS_cd_016
Test case group	SIP-SIP/Service/CD
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55D
Test purpose	SIP-I support. CD performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'. User A and user C are in Network A. User B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa. Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is deflected to user C; user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation allowed' and Redirection information if ISUP/BICC – SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Release diverting number information
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator presentation allowed Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation allowed Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason Deflection immediate or Deflection during alerting --[any boundary name]--

Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE(Call-ID A-B) → ← 180 Ringing (Call-ID B-A) in case CDa CD is performed ← INVITE(Call-ID B-C, IAM) Apply post test routine	
Comments	Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A Check: Is an INVITE request received at the interconnection interface? Check: Is an IAM encapsulated in the INVITE? Check: Is the Redirecting number present and is the Address presentation restricted indicator set to 'presentation allowed'? Check: Is the Original called number present and is the Address presentation restricted indicator set to 'presentation allowed'? Check: Is the Redirection number present? Check: Is Redirection information present and is the Redirecting reason 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 55D
Test purpose	SIP-I support. CD performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'. User A and user C are in Network A. User B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa. Served user releases his/her number to diverted-to user = Release diverting number information. Ensure that when user A calls user B, the call is deflected to user C; user C is notified of call diversion and informed of the diverting number. The notification information is present in the encapsulated IAM contained in the Redirecting number 'presentation restricted' and Redirection information if ISUP/BICC – SIP-I interworking is applicable in Network B.
Configuration	Subscription options: • Served user releases his/her number to diverted-to user = Do not release diverting number information
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Redirecting number Address presentation restricted indicator

	presentation restricted Address signal (*Diverting user*) Original called number Address presentation restricted indicator presentation restricted Address signal Redirection information Original Redirection Reason unknown Redirecting indicator Redirection counter Redirecting reason Deflection immediate or Deflection during alerting --[any boundary name]--
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 is the Address presentation restricted indicator set to 'presentation restricted'? Check: Is the Original called number present and is the Address presentation restricted indicator set to 'presentation restricted'? Check: Is the Redirection number present? Check: Is Redirection information present and is the Redirecting reason 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_multipleCFU_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. User A and user C are in Network A. User B and user D are in network B. User B and user C are provided with CFU "Served user allows the presentation of his/her URI to diverted-to user" = No.

<table>
<tr><td></td><td>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.</td></tr>
<tr><td>Configuration</td><td>Subscription options:
"Served user allows the presentation of his/her URI to diverted-to user" = No</td></tr>
<tr><td>SIP Parameter</td><td>INVITE:
History-Info:
<sip; userB@networkB?privacy=history >;index=1,
<sip:userC@networkA;cause=302>;index=1.1,

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</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
INVITE(Call-ID A-B) →
CFU is performed
← INVITE(Call-ID B-C)
CFU is performed
INVITE(Call-ID C-D) →
Apply post test routine</td></tr>
<tr><td>Comments</td><td>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 1 – The history entries can be accumulated in “one” History-Info header or each history entry is present in one single History-Info header.
NOTE 2 – The Request line may contain a ‘cause’ parameter indicating the redirecting reason.
Repeat this test in reverse direction.</td></tr>
</table>

<table>
<tr><td>Test case number</td><td>SS_ multipleCFU_002</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/multibleCF</td></tr>
<tr><td>Reference</td><td>4.5.2.6/[9]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>SE 17 AND SE 22</td></tr>
<tr><td>Test purpose</td><td>Call establishment with multiple forwarding.
User A and user C are in Network A. User B and user D are in network B. 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.</td></tr>
<tr><td>Configuration</td><td>Subscription options:
"Served user allows the presentation of his/her URI to diverted-to user" = Yes</td></tr>
<tr><td>SIP Parameter</td><td>INVITE:
History-Info header:
<sip:userB@networkB>;index=1,
<sip:userC@networkA;cause=302>;index=1.1,

INVITE:
History-Info header:
<sip:userB@networkB>;index=1,
<sip:userC@networkA;cause=302>;index=1.1,
<sip:userD@networkB;cause=302>;index=1.1.1</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
INVITE(Call-ID A-B) →
CFU is performed
← INVITE(Call-ID B-C)
CFU is performed
INVITE(Call-ID C-D) →
Apply post test routine</td></tr>
<tr><td>Comments</td><td>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?</td></tr>
</table>

	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 1 – The history entries can be accumulated in “one” History-Info header or each history entry is present in one single History-Info header. NOTE 2 – The Request line may contain a ‘cause’ parameter indicating the redirecting reason. Repeat this test in reverse direction.

7.1.5.7 Conference (CONF)

Test case number	SS_conf_001
Test case group	SIP-SIP/Service/CONF
Reference	4.5.2/[ETSI TS 124 605]
SELECTION EXPRESSION	([Network A] SE 11 AND [Network B] SE 11) AND SE 31
Test purpose	3Party 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, user B1 sends a NOTIFY to user A indicating 'Trying'. User B1 sends an INVITE request to the conference focus in Network A. If 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, user B2 sends a NOTIFY to user A indicating 'Trying'. User B2 sends an INVITE request to the conference focus in Network A. If the request is confirmed, user B2 sends a NOTIFY indicating '200 OK'. User A terminates the original dialogue.
Configuration	
SIP Parameter	REFER(user B1) Refer-To: <uri of conference focus;method=INVITE > NOTIFY(B1, 100) Content-Type: message/sipfrag SIP/2.0 100 INVITE: Request URI: uri of conference focus From: user B1 NOTIFY(B1, 200) Content-Type: message/sipfrag SIP/2.0 200 OK REFER(user B2)

	Refer-To: <uri of conference focus;method=INVITE > NOTIFY(B2, 100) Content-Type: message/sipfrag SIP/2.0 100 INVITE: Request URI: uri of conference focus From: user B2 NOTIFY(B2, 200) Content-Type: message/sipfrag SIP/2.0 200 OK
Message flow SIP (Network A) Interconnection Interface SIP (Network B) Establish a confirmed session to user B1 from Network A to Network B and put it on hold Establish a confirmed session to user B2 from Network A to Network B and put it on hold User A establishes a 3PTY conversation REFER(user B1) → ← 202 Accepted ← NOTIFY(B1, 100) 200 OK NOTIFY → ← INVITE(focus, user B1) 200 INVITE → ← ACK ← NOTIFY(B1, 200) 200 OK NOTIFY → BYE(user B1) → ← 200 OK BYE REFER(user B2) → ← 202 Accepted ← NOTIFY(100) 200 OK NOTIFY → ← INVITE(focus, user B2) 200 INVITE → ← ACK ← NOTIFY(B2, 200) 200 OK NOTIFY → BYE(user B2) → ← 200 OK BYE Apply post test routine	
Comments	User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on HOLD

	Check: The Refer-To header in the REFER method sent to user B1 and B2 contains the URI of the conference focus and is the method parameter set to 'INVITE' Check: The NOTIFY after the REFER request contains the 'SIP/2.0 100' message body. Check: The INVITE request is sent by user B1 and user B2 to the conference focus; the Request URI is used from the Refer-To header of the received REFER request Check: The NOTIFY after the REFER request contains the 'SIP/2.0 200 OK' message body. Check: The original session is terminated by user A. Repeat this test in reverse direction.

Test case number	SS_conf_002
Test case group	SIP-SIP/Service/CONF
Reference	4.5.2/[ETSI TS 124 605], 4.7.2.9.7/[ETSI TS 124 628]
SELECTION EXPRESSION	[Network A] SE 12 AND SE 31
Test purpose	3 Party establishment using reINVITE performed by the AS in Network A. User B1 and user B2 are located in Network B, user A is located in Network A. A confirmed session from user A to user B1 is set on hold; a confirmed session from user A to user B2 is set on hold. • Ensure that user A can invite user B1 to the conference by sending a reINVITE request. • Ensure that user A can invite user B2 to the conference by sending a reINVITE request.
Configuration	
SIP Parameter	INVITE <B1> From: <userA> To: <userB1> Call-ID: A-B1 P-Asserted-Identity: <userA> SDP: a=sendrecv INVITE <B2> From: <userA> Call-ID: A-B2 To: <userB2> P-Asserted-Identity: <userA> SDP: a=sendrecv
Message flow SIP (Network A) Interconnection Interface SIP (Network B) Establish a confirmed session to user B1 from Network A to Network B and put it on hold Establish a confirmed session to user B2 from Network A to Network B and put it on hold User A establishes a 3PTY conversation	

INVITE(Call-ID A-B1) → ← 200 INVITE ACK → INVITE(Call-ID A-B2) → ← 200 INVITE ACK → Apply post test routine	
Comments	User A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 are set on HOLD. Check: An INVITE is sent to user B1 and user B2 indicating a new IP address in the 'c' line of the SDP. Check: The 'a' line indicates 'sendrecv' Repeat this test in reverse direction.

Test case number	SS_conf_003
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	SIP-I/ISUP interworking. Served user establishes a 3 Party communication. Served User A is located in Network A and ISUP/BICC – SIP-I interworking applies in Network A. User A establishes a confirmed communication with a User B1 in Network B and sets it on HOLD. User A establishes a confirmed communication with a User B2 in Network B. Ensure that when User A establishes a 3 PTY communication • an INFO request is sent to User B1 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established' • an INFO request is sent to User B2 in Network B and a ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established'
Configuration	ISUP/BICC interworking applies in Network A User in Network A is subscribed to the 3PTY supplementary service
SIP Parameter	INFO <B1> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic Notification Conference established INFO <B2> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic Notification

	Conference established
Message flow SIP (Network A) Interconnection Interface SIP (Network B) Establish a confirmed session from User A in Network A to user B1 in Network B and put it on hold Establish a confirmed session from User A in Network A to user B2 in Network B INFO(Call-ID A-B1, CPG) → ← 200 INFO INFO(Call-ID A-B2, CPG) → ← 200 INFO Apply post test routine	
Comments	User A establishes confirmed communication to user B1 in Network B and sets it on hold. User A establishes a confirmed communication to user B2 in Network B. User A invokes the 3PTY communication. Check: Is an INFO request sent to user B1 and user B2 in Network B? Check: Is 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_004
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	SIP-I/ISUP interworking. Served user disconnects one of the remote users Served User A is located in Network A and ISUP/BICC – SIP-I interworking applies in Network A. User A establishes a confirmed communication with a User B1 in Network B and sets it on HOLD. User A establishes a confirmed communication with a User B2 in Network B. User A invokes 3PTY conversation. Ensure that when User A disconnects the previous active user • a BYE request is sent to User B1 in Network B • 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> 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 BYE(Call-ID A-B1, REL) → ← 200 INFO INFO(Call-ID A-B2, CPG) → ← 200 INFO Apply post test routine	
Comments	User A establishes a 3PTY conversation with user B1 and user B2 located in Network B. User A disconnects the communication with user B1 in Network B (previous on hold). Check: Is a BYE request sent to user B1 in Network B? Check: Is an ISUP/BICC CPG message encapsulated in the INFO request to user B2 in Network B? Check: Is the Generic Notification parameter in the encapsulated CPG in the INFO sent to user B2 set to 'Conference disconnected'? Repeat this test in reverse direction.

Test case number	SS_conf_005
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	SIP-I/ISUP interworking. Served user splits the 3 Party communication. Served User A is located in Network A and ISUP/BICC – SIP-I interworking applies in Network A. User A establishes a confirmed communication with a User B1 in Network B and sets it on HOLD. User A establishes a confirmed communication with a User B2 in Network B. User A invokes 3PTY conversation. Ensure that when User A splits the 3 PTY communication • an INFO request is sent to User B1 in Network B and 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.

<table>
<tr><td>SIP Parameter</td><td>INFO <B1>
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

CPG
Generic Notification
Conference disconnected

INFO <B2>
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required

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

INFO(Call-ID A-B2, CPG) →
← 200 INFO
Apply post test routine</td></tr>
<tr><td>Comments</td><td>User A establishes confirmed communication to user B1 in Network B and sets it on hold.
User A establishes a confirmed communication to user B2 in Network B.
Check: Is an INFO request sent to user B1 and user B2 in Network B?
Check: Is 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.</td></tr>
</table>

Test case number	SS_conf_006
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56

Test purpose	SIP-I/ISUP interworking. Establishment of a CONF conversation. Served User A is located in Network A and ISUP/BICC – SIP-I interworking applies in Network A. User A establishes a confirmed communication with a User B1 in Network B and invokes the CONF communication. Ensure that when User A invokes the CONF communication an INFO request is sent to User B1 in Network B and a ISUP/BICC CPG is encapsulated; the Generic Notification is set to 'conference established' when the conference is invoked. User A establishes a confirmed communication with a User B2 in Network B. Ensure when User A adds 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 'conference established' when the user is added to the conference.
Configuration	ISUP/BICC interworking applies in Network A. User in Network A is subscribed to the 3PTY supplementary service.
SIP Parameter	INFO1 <B1> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic Notification conference established INFO2 <B1> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic Notification Other party added INFO <B2> Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic Notification conference established

Message flow	
SIP (Network A) Interconnection Interface SIP (Network B) Establish a confirmed session from User A in Network A to user B1 in Network B User A establishes a CONF conversation INFO1(Call-ID A-B1, CPG) → ← 200 INFO Establish a confirmed session from User A in Network A to user B2 in Network B and add to the conference INFO2(Call-ID A-B2, CPG) → ← 200 INFO INFO(Call-ID A-B2, CPG) → ← 200 INFO Apply post test routine	
Comments	User A establishes confirmed communication to user B1 in Network B and invokes the CONF communication. Check: Is an INFO request sent to user B1 and in Network B and Is a ISUP/BICC CPG message encapsulated in the INFO request and the Generic Notification is set to 'conference established'? User A establishes a confirmed communication to user B2 in Network B and adds it to the conference. Check: Is an INFO request sent to user B2 Network B and a ISUP/BICC CPG message encapsulated the Generic Notification is set to 'conference established'? Check: Is an INFO request sent to user B1 Network B and a ISUP/BICC CPG message encapsulated the Generic Notification is set to 'Other party added'? Repeat this test in reverse direction.

Test case number	SS_conf_007
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56

Test purpose	SIP-I/ISUP interworking. Isolation and Reattachment of one party of the conference. Served User A is located in Network A and ISUP/BICC – SIP-I interworking applies in Network A. User A invokes a CONF communication with user B1 and user B2 in Network B. Ensure that when User A isolates one remote party (B1) from the CONF communication Ensure that when User A isolates one remote party (B1) from the CONF communication: • an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'isolated' in the encapsulated ISUP/BICCCPG. • an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party isolated' in the encapsulated ISUP/BICCCPG. • Ensure that when User A reattaches one remote party (B1) to the CONF communication • an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'reattached' in the encapsulated ISUP/BICCCPG. • an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party reattached' in the encapsulated ISUP/BICCCPG.
Configuration	ISUP/BICC interworking applies in Network A. User in Network A is subscribed to the 3PTY supplementary service.
SIP Parameter	INFO1 <B1> CPG Generic Notification= isolated INFO2 <B1> CPG Generic Notification= Other party isolated INFO1 <B2> CPG Generic Notification= reattached INFO2 <B2> CPG Generic Notification= Other party reattached
Message flow SIP (Network A) Interconnection Interface SIP (Network B) Establish a CONF communication with User B1 and User B2 in Network B User A isolates User B1 from the CONF conversation INFO1(Call-ID A-B1, CPG) → ← 200 INFO INFO1(Call-ID A-B2, CPG) → ← 200 INFO	

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

Test case number	SS_conf_008
Test case group	SIP-SIP/Service/CONF
Reference	5.4/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 56
Test purpose	SIP-I/ISUP interworking. Splitting and Adding of a party. Served User A is located in Network A and ISUP/BICC – SIP-I interworking applies in Network A. User A invokes a CONF communication with user B1 and user B2 in Network B. Ensure that when User A split one remote party (B1) from the CONF communication Ensure that when User A split one remote party (B1) from the CONF communication: • an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'conference disconnected' in the encapsulated ISUP/BICCCPG. • an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party split' in the encapsulated ISUP/BICCCPG. • Ensure that when User A adds one remote party (B1) to the CONF communication . • an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'Conference established' in the encapsulated ISUP/BICCCPG. • an INFO request is sent to User B2 in Network B and the Generic Notification is set to 'Other party added' in the encapsulated ISUP/BICCCPG.
Configuration	ISUP/BICC interworking applies in Network A. User in Network A is subscribed to the 3PTY supplementary service.

SIP Parameter	INFO1 <B1> CPG Generic Notification= conference disconnected INFO2 <B1> CPG Generic Notification=Other party split INFO3 <B2> CPG Generic Notification=Conference established INFO4 <B2> CPG Generic Notification= Other party added
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) Establish a CONF communication with User B1 and User B2 in Network B User A isolates User B1 from the CONF conversation INFO1(Call-ID A-B1, CPG) → ← 200 INFO INFO3(Call-ID A-B2, CPG) → ← 200 INFO User 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. User A splits user B1 in Network B from the CONF conversation. Check: Is an INFO request sent to user B1 and is the Generic notification set to 'conference disconnected' in the encapsulated CPG? Check: Is an INFO request sent to user B2 and is the Generic notification 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 is the Generic notification set to 'Conference established' in the encapsulated CPG? Check: Is an INFO request sent to user B2 and is the Generic notification set to 'Other party added' in the encapsulated CPG? Repeat this test in reverse direction.

7.1.5.8 Anonymous communication rejection (ACR) and communication barring (CB)

Test case number	SS_acr-cb_001
Test case group	SIP-SIP/Service/ACR-CB
Reference	4.5.2.6/[ETSI TS 124 611]

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

Test case number	SS_acr-cb_002
Test case group	SIP-SIP/Service/ACR-CB
Reference	4.5.2.6/[ETSI TS 124 611]
SELECTION EXPRESSION	SE 33
Test purpose	ACR performed in Network B for user B User A is located in Network A and user B is located in Network B and is subscribed to the Anonymous Communication rejection service. Ensure that an anonymous communication from user A is rejected in Network B by sending a 403 Anonymity Disallowed final response due to the Anonymous Communication Rejection service of user B.
Configuration	User B is subscribed to the Anonymous Communication Rejection service
SIP Parameter	INVITE P-Asserted-Identity: <URI of user A> Privacy: id
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE → ← 433 (Anonymity Disallowed) ACK →
Comments	Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the communication rejected by sending a 433 (Anonymity Disallowed) final response sent to user A? Repeat this test in reverse direction.

Test case number	SS_acr-cb_003
Test case group	SIP-SIP/Service/ACR-CB
Reference	6.5/[ITU-T Q.1912.5]
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 of user A> Privacy: id 433 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL: Cause indicator Cause = 21
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE ➔ ← 603 Decline (REL) ACK ➔	
Comments	Check: Is the P-Asserted-Identity present? Check: Is the Privacy header set to 'id'? Check: Is the communication rejected by sending a 603 Decline final response to user A? Check: Is an ISUP/BICC REL present in the 603 and is the cause value set to '21'? Repeat this test in reverse direction.

7.1.5.9 Closed user group (CUG)

Test case number	SS_cug_001
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.4/[ETSI TS 124 654]
SELECTION EXPRESSION	SE 34
Test purpose	Originating user +OA to terminating user no CUG An originating user in a CUG Outgoing Access allowed calls to a user not in a CUG. The session establishment is successful.

Configuration	Originating user: CUG, outgoing access allowed
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml Content-Disposition: signal;handling= ...... ……. <…cug> <…networkIndicator>01</… networkIndicator <…networkIndicator>23</… networkIndicator <….cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode> <…CugCommunicationIndicator>10</…cugCommunicationIndicator> <…:cug>
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing Apply post test routine	
Comments	Check: Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml? Check: Contains the XML body in the INVITE a 'cug' element? Check: Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element? Check: Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element? Check: Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '10' as a 'cug' child element? Check: Is the session setup not rejected? Repeat this test in reverse direction. NOTE – The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug_002
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.4, 4.5.2.10/[ETSI TS 124 654]
SELECTION EXPRESSION	SE 34
Test purpose	Originating user -OA to terminating user no CUG. An originating user in a CUG Outgoing Access not allowed calls to a user not in a CUG. The session establishment is not successful, a 403 (Forbidden) response is sent.
Configuration	Originating user: CUG, outgoing access not allowed
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml Content-Disposition: signal;handling= required …….

	<…:cug> <…networkIndicator>01</…networkIndicator <…networkIndicator>23</…networkIndicator <….ugInterlockBinaryCode>0F03</…cugInterlockBinaryCode> <…cugCommunicationIndicator>11</…cugCommunicationIndicator> <…cug>
Message flow SIP (Network A)	Interconnection Interface — SIP (Network B) INVITE → ← 403 (Forbidden) ACK →
Comments	Check: Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml? Check: Is the handling parameter in the Content-Disposition header set to required? Check: Contains the XML body in the INVITE a 'cug' element? Check: Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element? Check: Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element? Check: Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? Check: Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network. Repeat this test in reverse direction. NOTE – The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug_003
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.4, 4.5.2.10/[ETSI TS 124 654]
SELECTION EXPRESSION	SE 34
Test purpose	Originating user –OA to terminating user –IA. An originating user in a CUG Outgoing Access not allowed calls to a user in the same CUG. Incoming Access not allowed. The session establishment is successful.
Configuration	Originating user: CUG, outgoing access not allowed. Terminating user: CUG incoming access not allowed. User in Network A and user in Network B are in the same CUG.
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml Content-Disposition: signal;handling= required ……. <…cug>

	<…networkIndicator>01</…networkIndicator <…networkIndicator>23</…networkIndicator <…cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode> <…cugCommunicationIndicator>11</…cugCommunicationIndicator> <…cug>
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ 🡐 180 Ringing Apply post test routine	
Comments	Check: Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml? Check: Is the handling parameter in the Content-Disposition header set to required? Check: Contains the XML body in the INVITE a 'cug' element? Check: Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element? Check: Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element? Check: Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? Check: Is the session setup not rejected? Repeat this test in reverse direction. NOTE – The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug_004
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.4, 4.5.2.10/[ETSI TS 124 654]
SELECTION EXPRESSION	SE 34
Test purpose	Originating user in a CUG to terminating user –IA. An originating user in a CUG calls to a user in a different CUG. Incoming Access not allowed. The session establishment is not successful, a 403 (Forbidden) response is sent.
Configuration	User in Network A and user in Network B are not in the same CUG. Terminating user: CUG incoming access not allowed.
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml Content-Disposition: signal;handling= requiredv ……. <…cug> <…networkIndicator>01</…networkIndicator <…networkIndicator>23</…networkIndicator <…cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode>

	<…cugCommunicationIndicator>..</…cugCommunicationIndicator> <…cug>
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 403 (Forbidden) ACK →	
Comments	Check: Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml? Check: Contains the XML body in the INVITE a 'cug' element? Check: Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element? Check: Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element? Check: Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '10' or '11'as a 'cug' child element? Check: Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network Repeat this test in reverse direction. NOTE – The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug_005
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.10/[ETSI TS 124 654]
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/[ETSI TS 124 654]

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	SIP-SIP/Service/CUG
Reference	4.5.2.4/[ETSI TS 124 654]
SELECTION EXPRESSION	SE 34
Test purpose	Originating user –OA, Network B does not support CUG. An originating user in a CUG; Outgoing Access not allowed calls to a user in Network B. Network B does not support CUG. The session establishment is not successful, a 4xx unsuccessful final response is sent.
Configuration	
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml Content-Disposition: signal;handling= required ……. <…cug> <…networkIndicator>01</…networkIndicator <…networkIndicator>23</…networkIndicator <…cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode> <…cugCommunicationIndicator>11</…cugCommunicationIndicator> <…cug>
Message flow SIP (Network A)	Interconnection Interface — SIP (Network B) INVITE → ← 4xx/501 Not Implemented ACK →
Comments	Check: Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml?

	Check: Is the handling parameter in the Content-Disposition header set to required? Check: Contains the XML body in the INVITE a 'cug' element? Check: Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element? Check: Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element? Check: Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? Check: Is the session setup rejected by sending an unsuccessful final response? Repeat this test in reverse direction. NOTE – The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug_007A
Test case group	SIP-SIP/Service/CUG
Reference	4.5.2.4/[ETSI TS 124 654]
SELECTION EXPRESSION	SE 34
Test purpose	Originating user CUG-OA to terminating CUG user +ICB An originating user in a CUG outgoing access not allowed calls a user in the same CUG Incoming communication barred. The session establishment is not successful, a 603 (Decline) response is sent.
Configuration	User in Network B in a CUG incoming Communication Barring
SIP Parameter	INVITE: Content-Type: application/vnd.etsi.cug+xml Content-Disposition: signal;handling= required ……. <…cug> <…networkIndicator>01</…networkIndicator <…networkIndicator>23</…networkIndicator <…cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode> <…cugCommunicationIndicator>11</…cugCommunicationIndicator> <…cug>
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 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 this test in reverse direction. NOTE – The networkIndicator element value and the cugInterlockBinaryCode element value are examples.

Test case number	SS_cug_008
Test case group	SIP-SIP/Service/CUG
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 58
Test purpose	SIP-I/ISUP interworking. CUG call with outgoing access allowed User A is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. Ensure that when user A is in a CUG, 'outgoing access allowed' calls user B in Network B. The call is successful. There is 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) INVITE → ← 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, is the CUG Call Indicator 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 58
Test purpose	SIP-I/ISUP interworking. CUG call with outgoing access not allowed User A is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. Ensure that when user A is in a CUG, 'outgoing access allowed' calls user B in Network B. The call is successful. There is 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 --[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 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, is the CUG Call Indicator 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 58) AND ([Network B] SE 17 AND SE 47 AND SE 58)
Test purpose	SIP-I/ISUP interworking. CUG call with outgoing access not allowed (both users in the same CUG). User A in a CUG is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. User B is located in the PSTN/PLMN part and SIP-I – ISUP/BICC interworking applies in the same CUG. Ensure that when user A is in a CUG 'outgoing access not allowed' calls user B in Network B. The call is successful. There is 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. • 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 to Network B? Check: Is the Optional forward call indicator present, is the CUG Call Indicator set to 'Outgoing access not allowed'? Check: Is the CUG interlock code parameter present in the encapsulated IAM?

	Check: Is the call setup successful? Repeat this test in reverse direction.

Test case number	SS_cug_011
Test case group	SIP-SIP/Service/CUG
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 58) AND ([Network B] SE 17 AND SE 47 AND SE 58)
Test purpose	SIP-I/ISUP interworking. CUG calls to a CUG user incoming access not allowed (both user in the same CUG). User A in a CUG is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. User B is located in the PSTN/PLMN part and SIP-I – ISUP/BICC interworking applies in the same CUG. Ensure that when user A is in a CUG 'outgoing access not allowed' calls CUG user B in Network B. The call is successful. There is 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, is the CUG Call Indicator 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 58) AND ([Network B] SE 17 AND SE 47 AND SE 58)
Test purpose	SIP-I/ISUP interworking. CUG call to a CUG user incoming access not allowed (both user in different CUG). User A in a CUG is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. User B is located in the PSTN/PLMN part and SIP-I – ISUP/BICC interworking applies in different CUG. Ensure that when user A is in a CUG 'outgoing access not allowed' calls CUG user B in Network B. There is 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. A ISUP/BICC REL is encapsulated and the Cause value is set to '87'.
Configuration	• User in PSTN/PLMN part of Network A in a CUG outgoing access not allowed • User in PSTN/PLMN part of Network B in a CUG incoming access not allowed • User A and User B are in different CUG
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Optional Forward call indicator CUG Call Indicator Outgoing access not allowed CUG interlock code --[any boundary name]-- 500 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause indicators Cause value 87

Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 500 Server Internal error(REL) ACK →	
Comments	User A in the PSTN/PLMN part of Network A calls user B in Network B. User B in the PSTN/PLMN part of Network B. Check: Is an IAM encapsulated in the INVITE request sent from Network A to Network B? Check: Is the Optional forward call indicator present, is the CUG Call Indicator 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 is an ISUP/BICC REL encapsulated, and is the cause value 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 58
Test purpose	SIP-I/ISUP interworking. Call to a CUG user incoming access not allowed. User A is located in Network A. User B in a CUG Incoming access not allowed is located in the PSTN/PLMN part and SIP-I – ISUP/BICC interworking applies. Ensure that when user A calls user B in Network B, the call is rejected with a 500 (Server Internal error) final response. 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 is an ISUP/BICC REL encapsulated, and is the cause value set to 87? Repeat this test in reverse direction.

Test case number	SS_cug_014
Test case group	SIP-SIP/Service/CUG
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 58
Test purpose	SIP-I/ISUP interworking. Call to a CUG user incoming access allowed. User A is located in Network A. User B is located in the PSTN/PLMN part and SIP-I – ISUP/BICC interworking applied. Ensure that when user A calls CUG user B, incoming access allowed in Network B. The call is successful.
Configuration	User in PSTN/PLMN part of Network B in a CUG incoming access allowed
SIP Parameter	
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing 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.

7.1.5.10 Communication waiting (CW)

Test case number	SS_cw_001
Test case group	SIP-SIP/Service/CW
Reference	4.5.5.2/[b-ETSI TS 124 615]
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 user B is NDUB or UDUB.
Configuration	User B subscribed to the CW service
SIP Parameter	180: Alert-Info: <urn:alert:service:call-waiting>
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing Apply post test routine	

Comments	Check: Is an Alert-Info header present in the 180 Ringing Response and is the value set to '<urn:alert:service:call-waiting>'? Repeat this test in reverse direction.

Test case number	SS_cw_002
Test case group	SIP-SIP/Service/CW
Reference	4.5.5.2/[b-ETSI TS 124 615]
SELECTION EXPRESSION	SE 35 AND SE 36
Test purpose	Call rejected after timeout TAS-CW. User A is located in Network A, user B is located in Network B and subscribed to the communication Waiting service. Ensure that when user A calls user B, user A receives the 'communication Waiting indication' in the 180 Ringing provisional response if user B is NDUB or UDUB. After timeout TAS-CW Network B sends a 480 (Temporarily unavailable) response toward user A and the Reason header field is set to '19'.
Configuration	
SIP Parameter	180: Alert-Info: <urn:alert:service:call-waiting> 480: Reason: Q.850 ;cause=19
Message flow SIP (Network A)	Interconnection Interface — SIP (Network B) INVITE → ← 180 Ringing Timeout TAS-CW ← 480 (Temporarily unavailable) ACK →
Comments	Check: Is an Alert-Info header present in the 180 Ringing Response and is the value set to '<urn:alert:service:call-waiting>'? Check: Is a Reason header present in the 480 Response and is the protocol set to 'Q.850' and the cause parameter set to '19' Repeat this test in reverse direction.

Test case number	SS_cw_003
Test case group	SIP-SIP/Service/CW
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 59
Test purpose	SIP-I support. Call Waiting indication in 180 with encapsulated ACM. User A is located in Network A, user B is located in the PSTN/PLMN part of Network B and subscribed to the Call Waiting service. Ensure that when user A calls user B, an encapsulated ISUP/BICC ACM Generic notification 'call is a waiting call' is present in the 180 Ringing provisional response if user B is NDUB.

Configuration	User B subscribed to the CW service.
SIP Parameter	180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM Backward call indicator Called party's status indicator subscriber free Generic notification Notification indicator call is a waiting call
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing Apply post test routine	
Comments	Check: Is an ISUP/BICC ACM present in the 180 provisional response and the Generic notification is set to 'call is a waiting call'? Repeat this test in reverse direction.

Test case number	SS_cw_004
Test case group	SIP-SIP/Service/CW
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 59
Test purpose	SIP-I support. Call Waiting indication in 180 with encapsulated CPG. User A is located in Network A, user B is located in the PSTN/PLMN part of Network B and subscribed to the Call Waiting service. Ensure that when user A calls user B, an encapsulated ISUP/BICC CPG Generic notification 'call is a waiting call' is present in the 180 Ringing provisional response if user B is NDUB.
Configuration	User B subscribed to the CW service
SIP Parameter	180 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Event information Event indicator ALERTING Generic notification Notification indicator call is a waiting call

Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 183 Session Progress (ACM) ← 180 Ringing (CPG) Apply post test routine	
Comments	Check: Is an ISUP/BICC CPG present in the 180 provisional response and is the Generic notification set to 'call is a waiting call'? Repeat this test in reverse direction.

7.1.5.11 Explicit communication transfer (ECT)

Test case number	SS_ect_001
Test case group	SIP-SIP/Service/ECT
Reference	4.5.2/[ETSI TS 124 629]
SELECTION EXPRESSION	[Network A] SE 37 AND [Network A] SE 11 AND [Network A] SE 49
Test purpose	Blind/assured transfer using the REFER method. User A is located in Network A, user B and user C are located in Network B. User A invokes ECT to transfer a session with user B to user C. • Ensure that a REFER request is sent from Network A to Network B in the dialogue with user B. The URI in the Refer-To header is set to the address of the ECT AS in Network A and the method parameter is set to 'INVITE'. • Ensure that an INVITE request is sent from Network B to Network A and the Request URI is set to the address of the ECT AS in Network A. • Ensure that an INVITE request is sent from Network A to Network B and the Request URI is set to the address of user C.
Configuration	
SIP Parameter	REFER: Request URI address of user B Refer-To: <URI of ECT-AS>; method=invite INVITE1 Request URI address of ECT-AS INVITE2: Request URI address of user C
Message flow SIP (Network A) Interconnection Interface SIP (Network B) A confirmed session is established between user A and user B A confirmed session is established between user A and user C User A invokes ECT to transfer the session to user C REFER → ← 202 Accepted ← NOTIFY (100) 200 OK NOTIFY → CASE Blind transfer	

	BYE (A-B)	→
←	200 OK BYE	
←	INVITE1 (ECT-AS)	
	INVITE2 (user C)	→
←	200 OK INVITE	
	ACK	→
	200 OK INVITE	→
←	ACK	
←	NOTIFY (200)	
	200 OK NOTIFY	→
CASE Assured transfer		
	BYE (A-B)	→
←	200 OK BYE	
	Apply post test routine	

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

Test case number	SS_ect_002
Test case group	SIP-SIP/Service/ECT
Reference	4.5.2/[ETSI TS 124 629]
SELECTION EXPRESSION	[Network A] SE37 AND [Network A] SE 11 AND [Network A] SE 50
Test purpose	Consultative transfer using the REFER method. User A is located in Network A, user B and user C are located in Network B. User A invokes ECT to transfer a session with user B to user C. Ensure that a REFER request is sent from Network A to Network B in the dialogue with user B. The URI in the Refer-To header is set to the address of the ECT AS in Network A and the method parameter is set to 'INVITE'.

	Ensure that an INVITE request is sent from Network B to Network A and the Request URI is set to the address of the ECT AS in Network A. Ensure that an INVITE request is sent from Network A to Network B and the Request URI is set to the address of user C and a Replaces header is present containing the session identifiers of the session A – C.
Configuration	
SIP Parameter	REFER:Request URI address of user B Refer-To: <URI of ECT-AS>; method=invite INVITE1 Request URI address of ECT-AS INVITE2: Request URI address of user C Require: replaces Replaces: <session A-C>

Message flow

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

← BYE (A-C) 200 OK BYE → Apply post test routine	
Comments	Check: Is a REFER request sent to Network B, is the Refer-To header set to the URI of the ECT-AS in Network A and is a method parameter present set to 'INVITE'? Check: Is an INVITE request sent to Network A; is the Request line set to the address of the ECT-AS in Network A? Check: Is an INVITE request sent to Network B; is the Request set to the address of user C and does a Replaces header present contain 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/[ETSI TS 124 629], 4.7.2.9.7/[ETSI TS 124 628]
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 an user C are located in Network B User A invokes ECT to transfer a session with user B to user C. • Ensure that Network A establishes a session to user C. • Ensure that 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 INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes]
Message flow SIP (Network A) Interconnection Interface SIP (Network B) A confirmed session is established between user A and user B User A invokes ECT to transfer the session to user C INVITE1 (user C) → ← 180 Ringing ← 200 OK INVITE ACK → INVITE2 (user B) →	

← 200 OK INVITE ACK → Apply post test routine	
Comments	Check: Is an initial INVITE sent from Network A to user C to establish a dialogue between Network A and user C? Check: Is a reINVITE sent from Network A to user B update the session parameter in the SDP? Repeat this test in reverse direction.

Test case number	SS_ect_004
Test case group	SIP-SIP/Service/ECT
Reference	4.5.2/[ETSI TS 124 629], 4.7.2.9.7/[ETSI TS 124 628]
SELECTION EXPRESSION	[Network A] SE37 AND [Network A] SE 12 AND [Network A] SE 50
Test purpose	Consultative transfer using the 3pcc method. User A is located in Network A, user B and user C are located in Network B. User A invokes ECT to transfer a session with user B to user C. • Ensure that Network A sends a reINVITE to update the session between user A and user B (SDP: IP address, port and codec) • Ensure that Network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec)
Configuration	
SIP Parameter	INVITE1: Request URI address of user C SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes] INVITE2: Request URI address of user B SDP c=IN IP4/6 [new IP address] m=audio [new port] RTP/AVP [new codec list] a=[new attributes]
Message flow SIP (Network A) Interconnection Interface SIP (Network B) A confirmed session is established between user A and user B A confirmed session is established between user A and user C User A invokes ECT to transfer the session to user C INVITE1 (user B) → ← 200 OK INVITE ACK → INVITE2 (user C) → ← 200 OK INVITE ACK → Apply post test routine	

Comments	Check: Is a reINVITE is sent from Network A to user B update the session parameter in the SDP? Check: Is a reINVITE is sent from Network A to user C update the session parameter in the SDP? Repeat this test in reverse direction.

Test case number	SS_ect_005
Test case group	SIP-SIP/Service/ECT
Reference	5.4.3.2/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 60
Test purpose	SIP-I support. Call Transfer invoked in active state, call was previous on HOLD BICC/ISUP – SIP-I interworking applies in the originating network. User A and C are located in Network A and user B is located in Network B. Ensure that a User A can successfully invoke the ECT supplementary service and transfer the call with User B to User C in active state.
Configuration	User A is subscribed to the Explicit Call Transfer supplementary service
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/sdp a=sendrecv --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FAC Generic Notification Call transfer active Call transfer number --[any boundary name]--

Message flow

SIP (Network A)	Interconnection Interface	SIP (Network B)
	A confirmed session is established between user A and user B and set on hold	
	User A invokes ECT to transfer the session to user C	
	INFO (LOP request) →	
	← *200 OK INFO*	
	← *INFO (LOP response)*	
	200 OK INFO →	
CASE A		
	INVITE (sendrecv; FAC) →	
	← 200 OK INVITE	
	ACK →	

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

Test case number	SS_ect_006
Test case group	SIP-SIP/Service/ECT
Reference	5.4.3.2/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 60
Test purpose	SIP-I support. Call Transfer invoked in alerting state, call was previous on HOLD BICC/ISUP – SIP-I interworking applies in the originating network. User A and C are located in Network A and user B is located in Network B. Ensure that User A can successfully invoke the ECT supplementary service and transfer the call with User B to User C in alerting state.
Configuration	User A is subscribed to the Explicit Call Transfer supplementary service

<table>
<tr><td>SIP Parameter</td><td>INVITE
Content-Type: multipart/mixed;boundary=[any boundary name]
--[any boundary name]
Content-Type: application/sdp
a=sendrecv
--[any boundary name]
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
FAC
Generic Notification
Call transfer alerting
Call transfer number
--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow
SIP (Network A) Interconnection Interface SIP (Network B)
A confirmed session is established between user A and user B and set on hold
User A invokes ECT to transfer the session to user C
INFO (LOP request) →
← 200 OK INFO
← INFO (LOP response)
200 OK INFO →
CASE A
INVITE (sendrecv; FAC) →
← 200 OK INVITE
ACK →
CASE B
INFO (FAC) →
← 200 OK INFO
INVITE (sendrecv) →
← 200 OK INVITE
ACK →
Apply post test routine</td></tr>
</table>

Comments	A session from User A to User B is already established. User A sets User B on hold. A session from User A to User C is already established. User A invokes the ECT service. Check: Is (optional) an INFO request sent from Network A to Network B and is an ISUP LOP message present and the Loop prevention indicator set to 'request'? Check: Is (optional) an INFO request sent from Network A to Network B, and is an ISUP LOP message present and the Loop prevention indicator set to 'response'? Check: Is (CASE A) an INVITE request sent and is an ISUP FAC message present containing a Generic notification indicator set to 'Call transfer alerting' and, in addition, is the media stream set to 'sendrecv'? Check: Is (CASE B) an INFO request sent and is an ISUP FAC message present containing a Generic notification indicator set to 'Call transfer alerting'? In addition is an INVITE request sent and is the media stream set to 'sendrecv' to resume the held session? NOTE – The content of the FAC in the INVITE request is Equal to the content of the FAC in the INFO request. Repeat this test in reverse direction.

Test case number	SS_ect_007
Test case group	SIP-SIP/Service/ECT
Reference	5.4.3.2/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 60
Test purpose	SIP-I support. Call Transfer invoked in active state. BICC/ISUP – SIP-I interworking applies in the originating network. User A and B are located in Network A and User C is located in Network B. Ensure that User A can successfully invoke the ECT supplementary service and transfer the call with User B to User C in active state.
Configuration	User A is subscribed to the Explicit Call Transfer supplementary service
SIP Parameter	INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FAC Generic Notification Call transfer active Call transfer number

Message flow	
SIP (Network A) — Interconnection Interface — SIP (Network B) A confirmed session is established between user A and user C User A invokes ECT to transfer the session to user C *INFO (LOP request)* → ← *200 OK INFO* ← *INFO (LOP response)* *200 OK INFO* → INFO (FAC) → ← 200 OK INFO Apply post test routine	
Comments	A session from User A to User B is already established. User A sets user B on hold. A session from User A to User C is already established. User A invokes the ECT service. Check: Is (optional) an INFO request sent from Network A to Network B and an ISUP LOP message present, and the Loop prevention indicator set to 'request'? Check: Is (optional) an INFO request sent from Network A to Network B and an ISUP LOP message present and the Loop prevention indicator set to 'response'? Check: Is (CASE B) an INFO request sent and an ISUP FAC message present containing a Generic notification indicator set to 'Call transfer active'? NOTE – The content of the FAC in the INVITE request is Equal to the content of the FAC in the INFO request. Repeat this test in reverse direction.

Test case number	SS_ect_008
Test case group	SIP-SIP/Service/ECT
Reference	5.4.3.2/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 60
Test purpose	SIP-I support. Call Transfer invoked in alerting state. BICC/ISUP – SIP-I interworking applies in the originating network. User A and B are located in Network A and user C is located in Network B. Ensure that User A can successfully invoke the ECT supplementary service and transfer the call with User B to User C in alerting state.
Configuration	User A is subscribed to the Explicit Call Transfer supplementary service
SIP Parameter	INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required CPG Generic Notification Call transfer alerting Call transfer number

Message flow	
SIP (Network A) — Interconnection Interface — SIP (Network B) A session in the early dialogue is established between user A and user C User A invokes ECT to transfer the session to user C *INFO (LOP request)* → ← *200 OK INFO* ← *INFO (LOP response)* *200 OK INFO* → INFO (CPG) → ← 200 OK INFO Apply post test routine	
Comments	A session from User A to User B is already established. User A sets user B on hold. A session from User A to User C is already established. User A invokes the ECT service. Check: Is (optional) an INFO request sent from Network A to Network B and is an ISUP LOP message present and the Loop prevention indicator set to 'request'? Check: Is (optional) an INFO request sent from Network A to Network B and an ISUP LOP message present and the Loop prevention indicator set to 'response'? Check: Is (CASE B) an INFO request sent and an ISUP CPG message present containing a Generic notification indicator set to 'Call transfer alerting'? NOTE – The content of the FAC in the INVITE request is Equal to the content of the FAC in the INFO request. Repeat this test in reverse direction.

7.1.5.12 Malicious communication identification (MCID)

Test case number	SS_mcid_001
Test case group	SIP-SIP/Service/MCID
Reference	4.5.2.5/[ETSI TS 124 616]
SELECTION EXPRESSION	SE 38
Test purpose	Network B sends an MCID request, no response. User A is located in Network A, user B is located in Network B and subscribed to the Malicious Communication Identification service. When user A calls user B and no originating identification is present in the INVITE request, Network B sends an INFO request to Network A requesting the originating identity. After timeout of timer TO-ID, Network B sends the 180 Ringing response.
Configuration	User B is subscribed to the MCID service

SIP Parameter	INFO: <…:mcid……………....> <…:request> <…:McidRequestIndicator>01</…:McidRequestIndicator> <…:HoldingIndicator >…</…:HoldingIndicator> </…:request> </…:mcid>
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE → ← INFO 200 OK INFO → Timeout $T_{O\text{-}ID}$ ← 180 Ringing Apply post test routine	
Comments	Check: Is an INFO request sent to Network A? Check: Is the McidRequestIndicator element set to, 01'? Check: Is a 200 OK INFO response sent to Network B? Repeat this test in reverse direction.

Test case number	SS_mcid_002
Test case group	SIP-SIP/Service/MCID
Reference	4.5.2.5/[ETSI TS 124 616]
SELECTION EXPRESSION	[Network A] SE 38 AND [Network A] SE 38
Test purpose	Network B sends an 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, Network B sends an INFO request to Network B requesting the originating identity. After receipt of an INFO request from Network A, Network B sends the 180 Ringing response.
Configuration	User B subscribed to the MCID service User A is an ISDN or POTS user in the PSTN of Network A
SIP Parameter	INFO: <…:mcid ……………….....> <…:request> <…:McidRequestIndicator>01</…:McidRequestIndicator> <…:HoldingIndicator >…</…:HoldingIndicator> </…:request> </…:mcid> INFO: <…:mcid……………......> <…:response> <…:McidResponseIndicator>01</…:McidResponseIndicator>

	<…:HoldingProvidedIndicator>…</…:HoldingProvidedIndicator> <…:OrigPartyIdentity>any URI</…:OrigPartyIdentity> <…:OrigPartyPresentationRestriction> true/false </…:OrigPartyPresentationRestriction> </…:response> </…:mcid>
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) INVITE → ← INFO 200 OK INFO → INFO → ← 200 OK INFO ← 180 Ringing Apply post test routine	
Comments	Check: Is an INFO request sent to Network A? Check: Is the McidRequestIndicator element set to ‚01'? Check: Is a 200 OK INFO response sent to Network B? Check: Is an INFO request sent to Network B? Check: Is the McidResponseIndicator element set to‚ 01'? Check: Is the OrigPartyIdentity element present in the response element? Check: Is a 200 OK INFO response sent to Network A? 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/[ITU-T Q.1912.5]
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 calls user B, and no originating identification is present in the INVITE request, Network B sends a 183 Session Progress request to Network A and an ISUP/BICC IDR message is present, the MCID request indicator is set to 'MCID requested' requesting the originating identity. After timeout of timer (ISUP) T39, 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 IDR MCID request indicators MCID request indicator MCID requested
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 183 Session Progress (IDR) Timeout T39 ← 180 Ringing (ACM) Apply post test routine	
Comments	Check: Is an 183 Session Progress sent to Network A? Check: Is an ISUP/BICC IDR message present and is the MCID request indicator 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/[ITU-T Q.1912.5]
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 an 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 calls user B and no originating identification is present in the INVITE request, Network B sends an 183 Session Progress to Network A requesting the originating identity. After receipt of an INFO request containing the ISUP IRS message from Network A, Network B sends the 180 Ringing response.
Configuration	User B subscribed to the MCID service. User A is an 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 Sesion Progress(IDR) INFO(IRS) → ← 200 OK INFO ← 180 Ringing(ACM) Apply post test routine	
Comments	Check: Is an 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 is an ISUP/BICC IRS present and is the MCID response indicator set to 'MCID included'? Check: Is the Calling party number present in the attached ISUP/BICC IRS? Check: Is a 200 OK INFO response sent to Network A? Repeat this test in reverse direction.

7.1.5.13 Message waiting indication (MWI)

Test case number	SS_mwi_001
Test case group	SIP-SIP/Service/MWI
Reference	4.7.2/[ETSI TS 124 606]
SELECTION EXPRESSION	[Network A] SE 39 AND [Network B] SE 39
Test purpose	Initial subscription of a Voicemail box. The Voicemail owner is in Network A, his Voicemail box is located in Network B. Ensure that a Voicemail owner is able to activate his Voicemail box.
Configuration	Voicemail in Network B Voicemail owner in Network A
SIP Parameter	SUBCRIBE

	Event: message-summary Expires: [any value] Accept: application/simple-message-summary NOTIFY Subscription-State: active;expires=[any value] Event: message-summary
Message flow SIP (Network A) Interconnection Interface SIP (Network B) SUBCRIBE → ← 200 OK SUBSCRIBE ← NOTIFY 200 OK NOTIFY → ← 200 OK BYE ← NOTIFY 200 OK NOTIFY → Apply post test routine	
Comments	Check: Is it possible for a user in Network A to subscribe to a voicemail box in Network B? Check: Is the Event header in the SUBCRIBE set to 'message-summary'? Check: Is the Accept header in the SUBCRIBE set to 'application/simple-message-summary'? Check: Is the Event header in the NOTIFY set to 'message-summary'? Repeat this test in reverse direction.

Test case number	SS_mwi_002
Test case group	SIP-SIP/Service/MWI
Reference	4.7.2/[ETSI TS 124 606]
SELECTION EXPRESSION	[Network A] SE 39 AND [Network B] SE 39
Test purpose	A new entry in the voicemail box is indicated to the owner. The voicemail owner is in Network A, his voicemail box is located in Network B. Ensure when a user calls user A and the call is not answered, the call is forwarded to the voicemail box of user A in Network B. Ensure that user A is notified by message waiting indication that there is a new message present in his voicemail account.
Configuration	Voicemail in Network B Voicemail owner in Network A

SIP Parameter	NOTIFY Subscription-State: active;expires=[any value] Event: message-summary Content-Type: application/simple-message-summary Messages-Waiting: yes Message-Account: sip:userA@networkA (optional) Voice-Message: [any new value]/[any old value] (optional)

Message flow

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

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

7.1.5.14 Completion of communications to busy subscriber (CCBS), completion of communications by no reply (CCNR)

Test case number	SS_cc_001
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.3/[ETSI TS 124 642]
SELECTION EXPRESSION	[Network B] SE 40
Test purpose	Indicating that CCBS is 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, that Network B sends an indication that CCBS is possible in the 486 Busy Here final response.
Configuration	
SIP Parameter	486: Call-Info: <sip:UE-B>;purpose=call-completion;m=BS

Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 486 Busy Here ACK →	
Comments	Check: The 486 final response contains the Call-Info header. Check: The Call-Info header contains the URI of user B as the monitor point in Network B. Check: The Call-Info header contains the purpose parameter set to 'call-completion' and the m parameter set to 'BS'. Repeat this test in reverse direction.

Test case number	SS_cc_002
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.3/[ETSI TS 124 642]
SELECTION EXPRESSION	[Network B] SE 41
Test purpose	Indicating that CCNR is 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, that Network B sends an indication that CCNR is possible in the 180 Ringing provisional response.
Configuration	
SIP Parameter	180: Call-Info: <sip:UE-B>;purpose=call-completion;m=NR
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing Apply post test routine	
Comments	Check: The 180 provisional response contains the Call-Info header. Check: The Call-Info header contains the URI of user B as the monitor point in Network B Check: The Call-Info header contains the purpose parameter set to 'call-completion' and the m parameter set to 'NR'. Repeat this test in reverse direction.

Test case number	SS_cc_003
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.2/[ETSI TS 124 642]
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 calls user B, and user B is busy, that the indication that CCBS is possible is sent to Network A. When user A invokes CCBS, a SUBSCRIBE request is sent to Network B, the Event header is set to 'call-completion' and the m parameter in the Request line is set to 'BS'. • Ensure when user A call user B and user B is free, the indication that CCNR is possible is sent to Network A. when user A invokes CCNR, a SUBSCRIBE request is sent to Network B, the Event header is set to 'call-completion' and the m parameter in the Request line is set to 'NR'. • Ensure that Network B sends a NOTIFY request to Network A to confirm that the request is in the Call completion queue at the terminating Application Server.
Configuration	
SIP Parameter	SUBSRIBE sip:B-AS;m=BS or m=NR From:<UE-A> To:<UE-B> Contact:<A-AS> Event:call-completion NOTIFY sip:A-AS Event:call-completion Content-Type: application/call-completion state: queued service-retention
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) An indication whether CCBS or CCNR is possible is sent by Network B SUBSCRIBE → ← 202 Accepted ← NOTIFY 200 OK NOTIFY → Apply post test routine	
Comments	Check: Is a SUBCRIBE request sent to Network B? Check: Is the m parameter in the Request URI set to 'BS' in case of CCBS request or set to 'NR' in case of CCNR? Check: Is a NOTIFY request sent to Network A and is the Event header set to 'call-completion' and is the state header in the message body set to 'queued''? Repeat this test in reverse direction. NOTE – The service-retention header in the NOTIFY body is a network option.

Test case number	SS_cc_004
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.3/[ETSI TS 124 642]
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 Network B is currently unable to process the request (e.g., the B-queue is full), a 480 Temporally Unavailable final response is sent.
Configuration	
SIP Parameter	SUBSRIBE sip:B-AS;m=BS or m=NR From:<UE-A> To:<UE-B> Contact:<A-AS> Event: call-completion
Message flow SIP (Network A) Interconnection Interface SIP (Network B) An indication whether CCBS or CCNR is possible is sent by Network B SUBSCRIBE ➔ 🡐 480 (Temporarily Unavailable)	
Comments	Check: Is a SUBCRIBE request sent to Network B? Check: Is the m parameter in the Request URI 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/[ETSI TS 124 642]
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 that when user B becomes available for CC recall, the CC recall procedure is started. 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 sent to Network A to indicate the subscription is terminated; the reason header is set to 'noresource'.
Configuration	

SIP Parameter	NOTIFY sip:O-AS Event: call-completion Content-Type: application/call-completion state: ready NOTIFY sip:O-AS Event: call-completion Subscription-State: terminated; reason=noresource
Message flow SIP (Network A) Interconnection Interface SIP (Network B) A CCBS or CCNR request was already successful ← NOTIFY 200 OK NOTIFY → INVITE → ← 180 Ringing ← NOTIFY 200 OK NOTIFY → ← 200 OK INVITE ACK → Apply post test routine	
Comments	Check: Is a NOTIFY request sent to Network A and is the Event header set to 'call-completion' and is the state header in the message body set to 'ready'? Check: Is the recall from user A to user B successful? Check: Is the CC revocation performed after the 180 Ringing or the 200 OK INVITE was sent to user A? Repeat this test in reverse direction.

Test case number	SS_cc_006
Test case group	SIP-SIP/Service/CC
Reference	4.5.4.31/[ETSI TS 124 642]
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 call to user B) 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 sent to Network A and is the Event header set to 'call-completion' and is the state header in the message body set to 'ready'? User A does not perform the recall. Check: Is the CC revocation 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/[ETSI TS 124 642]
SELECTION EXPRESSION	([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR [Network B] SE 41)
Test purpose	User A is unavailable while CC recall is performed. User A is located in Network A and user B is located in Network B. User A has successfully invoked a CCBS or CCNR request. User B is available for CC recall and Network B sends a CC-recall notification to Network A. • Ensure that Network A sends PUBLISH request to suspend the recall procedure • Ensure that Network A sends PUBLISH request to resume the recall procedure if user A is available to complete the recall procedure • Ensure the Network B sends a NOTIFY request to indicate the CC-recall procedure.
Configuration	

<table>
<tr><td>SIP Parameter</td><td>NOTIFY sip:O-AS
Event:call-completion
Content-Type: application/call-completion
state: ready

PUBLISH sip B-AS
To: SIP 2
Event: presence
Content-Type: application/pidf+xml
<?xml version="1.0" encoding="UTF-8"?>
<presence
<status>
<basic>closed</basic>

PUBLISH sip B-AS
To: SIP 2
Event: presence
Content-Type: application/pidf+xml
<?xml version="1.0" encoding="UTF-8"?>
<presence
<status>
<basic>open</basic></td></tr>
<tr><td colspan="2">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</td></tr>
<tr><td>Comments</td><td>Check: Is a PUBLISH request sent from Network A to Network B containg a "presence" XML element and the "basic" element is set to "closed"
Check: After 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.</td></tr>
</table>

Test case number	SS_cc_008
Test case group	SIP-SIP/Service/CC

Reference	6.11.2/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	SIP-I support: Indicating that CCBS possible BICC/ISUP – SIP-I interworking applies in the terminating network and User A is located in Network A and user B is located in Network B. Ensure when user A calls user B and user B is busy, that Network B sends a 486 Busy Here final response and an encapsulated ISUP REL is present, the Cause value indicator is set to #17 or #34 and the CCBS possible indicator is set to 'CCBS possible'.
Configuration	
SIP Parameter	486: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value #17 or #34 Diagnostics CCBS possible
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 486 Busy Here (REL) ACK →	
Comments	Check: The 486 final response contains an encapsulated BICC/ISUP REL, the Cause value set to 17 or 34 and the Diagnostics set to 'CCBS possible'. Repeat this test in reverse direction.

Test case number	SS_cc_009
Test case group	SIP-SIP/Service/CC
Reference	6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47
Test purpose	SIP-I support: Indicating that 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, that Network B sends a 180 Ringing provisional response and an encapsulated ACM is present containing a CCNR possible indicator set to 'CCNR possible'.
Configuration	
SIP Parameter	180: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM

	CCNR possible indicator CCNR possible
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing (ACM) Apply post test routine	
Comments	Check: The 180 provisional response contains an encapsulated ACM. Check: The CCNR possible indicator in the ACM is set to 'CCNR possible'. Repeat this test in reverse direction.

7.1.6 Other PSTN services (SIP-I interworking)

7.1.6.1 User-to-user Signalling (UUS)

Test case number	SS_uus_001
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 63
Test purpose	SIP-I support: Indicating of User-to-User service 1 implicit in initial INVITE request. BICC/ISUP – SIP-I interworking applies in the originating network. User A is located in Network A and user B is located in Network B. Ensure that 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 INVITE request.
Configuration	User A is subscribed to the User-to-User service 1 implicit request
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM User-to-user Information User Information
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE (IAM) → Apply post test routine	
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request? Check: Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM? Repeat this test in reverse direction.

Test case number	SS_uus_002
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1, 6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 63) AND ([Network B] SE 17 AND SE 47 AND SE 63)
Test purpose	SIP-I support: Indicating of User-to-User service 1 implicit response in 180. BICC/ISUP – SIP-I interworking applies in the originating and terminating network. User A is located in Network A and user B is located in Network B. Ensure that 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 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ACM User-to-user Information User Information
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE (IAM) → ← 180 Ringing (ACM) Apply post test routine	
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request? Check: Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM? Check: Is an ISUP/BICC ACM encapsulated in the 180 response? Check: Is a User-to-user Information parameter present in the encapsulated ISUP/BICC ACM? Repeat this test in reverse direction.

Test case number	SS_uus_003
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 63

Test purpose	SIP-I support: Indicating of User-to-User service 1 explicit in initial INVITE request BICC/ISUP – SIP-I interworking applies in the originating network. User A is located in Network A and user B is located in Network B. Ensure that 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 INVITE 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 63) AND ([Network B] SE 17 AND SE 47AND 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.

Test case number	SS_uus_005
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1, 6.5/[ITU-T Q.1912.5]
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 and 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47) AND ([Network B] SE 17 AND SE 47) AND SE 63
Test purpose	SIP-I support: Indicating of User-to-User service 1 essential explicit rejection BICC/ISUP – SIP-I interworking applies in the originating and terminating network. User A is located in Network A and user B is located in Network B. Ensure when user A subscribed to the User-to-User service 1 explicit request calls user B subscribed to User-to-User service 1 essential is rejected by the network or by the user. A 500 Server Internal Error is sent and an encapsulated ISUP/BICC REL is present, the Cause value is set to #29 or #69.
Configuration	User A is subscribed to the User-to-User service 1 explicit request

SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM User-to-user Indicator Request service 1 essential 500 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value #29 or #69
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE (IAM) → ← 500 Server Internal Error (REL) ACK → Apply post test routine	
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request? Check: Is a User-to-user Indicator parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 1', 'essential'? Check: Is an ISUP/BICC REL encapsulated in the 500 response? Check: Is the Cause value set to #29 or #69 in the encapsulated REL? Repeat this test in reverse direction.

Test case number	SS_uus_007
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 63
Test purpose	SIP-I support: Indicating of User-to-User service 2 in initial INVITE request. BICC/ISUP – SIP-I interworking applies in the originating network. User A is located in Network A and user B is located in Network B. Ensure that when user A, subscribed to the User-to-User service 2, calls user B, a User-to-user Indicator parameter is present and set to 'Request service 2', 'not essential' or 'essential' in the encapsulated IAM of the initial INVTE request.
Configuration	User A is subscribed to the User-to-User service 2

SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM User-to-user Indicator Request service 2 ‘not essential’ or 'essential'
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE (IAM) ➔ Apply post test routine	
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request and a User-to-user Indicator parameter is present set toRequest service 2 'not essential' or 'essential'? Repeat this test in reverse direction.

Test case number	SS_uus_008
Test case group	SIP-SIP/SIP-I/UUS
Reference	5.4.3.2, 6.5, 7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 63) AND ([Network B] SE 17 AND SE 47 AND SE 63)
Test purpose	SIP-I support: Indicating of User-to-User service 2 in initial INVITE request successful. BICC/ISUP – SIP-I interworking applies in the originating network. User A is located in Network A and user B is located in Network B. Ensure when user A, subscribed to the User-to-User service 2, calls user B, 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) ➔ ← 200 OK INFO ← 183 Session Progress (USR) Apply post test routine	
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request and is the User-to-user Indicator parameter is present set to Request service 2 'not essential' or 'essential"?" Check: Is an ISUP/BICC ACM encapsulated in the 180 and is the User-to-user Indicator parameter is present set to 'Response', 'service 2 provided'? Check: Is an ISUP/BICC USR encapsulated in the INFO message sent from Network A to Network B and does it contain a User-to-user Information parameter? Check: Is an ISUP/BICC USR encapsulated in the 183 response sent from Network B to Network A and does it contain a User-to-User Information parameter? Repeat this test in reverse direction.

Test case number	SS_uus_009
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1, 6.5/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 63) AND ([Network B] SE 17 AND SE 47 AND SE 63)

Test purpose	SIP-I support: Indicating of User-to-User service 2 not essential rejected in 180 response, BICC/ISUP – SIP-I interworking applies in the originating and terminating network. User A is located in Network A and user B is located in Network B. Ensure when user A, subscribed to the User-to-User service 2 not essential, calls user B, not subscribed to User-to-User service 2, the call is rejected by the network and 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
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE (IAM) → ← 180 Ringing (ACM) Apply post test routine	
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request? Check: Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 2' 'not essential'? Check: Is an ISUP/BICC ACM encapsulated in the 180 response? Check: Is a User-to-user Indicator parameter present set to 'Response', 'service 2 not provided' in the encapsulated ISUP/BICC ACM Repeat this test in reverse direction.

Test case number	SS_uus_010
Test case group	SIP-SIP/SIP-I/UUS
Reference	6.11.2, 7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 63) AND ([Network B] SE 17 AND SE 47 AND SE 63)

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

Test case number	SS_uus_011
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 63

Test purpose	SIP-I support: Indicating of User-to-User service 3 in initial INVITE request BICC/ISUP – SIP-I interworking applies in the originating network. User A is located in Network A and user B is located in Network B. Ensure when user A, subscribed to the User-to-User service 3, calls user B, 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 and the User-to-user Indicator parameter is present set to "Is the Request service 3 'not essential' or 'essential'?" Repeat this test in reverse direction.

Test case number	SS_uus_012
Test case group	SIP-SIP/SIP-I/UUS
Reference	5.4.3.2, 6.5, 7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 63) AND ([Network B] SE 17 AND SE 47 AND SE 63)
Test purpose	SIP-I support: Indicating of User-to-User service 3 in initial INVITE request successful. BICC/ISUP – SIP-I interworking applies in the originating network. User A is located in Network A and user B is located in Network B. Ensure when user A, subscribed to the User-to-User service 3, calls user B, 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. The User-to-User service is successful.
Configuration	User A is subscribed to the User-to-User service 3 User B is subscribed to the User-to-User service 3

<table>
<tr><td>SIP Parameter</td><td>INVITE:
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
IAM
User-to-user Indicator
Request
service 3
not essential or 'essential'

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

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

Test case number	SS_uus_013
Test case group	SIP-SIP/SIP-I/UUS
Reference	7.1, 6.5/[ITU-T Q.1912.50]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 63) AND ([Network B] SE 17 AND SE 47 AND SE 63)
Test purpose	SIP-I support: Indicating of User-to-User service 3 not essential rejected in 200 OK response BICC/ISUP – SIP-I interworking applies in the originating and terminating network. User A is located in Network A and user B is located in Network B. Ensure that 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 and set to 'Response', 'service 3 not provided' in the encapsulated ANM of the 200 OK final response.
Configuration	User A is subscribed to the User-to-User service 3 User B is not subscribed to the User-to-User service 3
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM User-to-user Indicator Request service 3 not essential 200 OK Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM User-to-user Indicator Response service 3 not provided
Message flow SIP (Network A) / Interconnection Interface / SIP (Network B) INVITE (IAM) → ← 180 Ringing (ACM) ← 200 OK INVITE (ANM) ACK → Apply post test routine	
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request? Check: Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 3' 'not essential'? Check: Is an ISUP/BICC ANM encapsulated in the 200 OK response? Check: Is 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 63) AND ([Network B] SE 17 AND SE 47 AND SE 63)
Test purpose	SIP-I support: Indicating of User-to-User service 3 essential rejection. BICC/ISUP – SIP-I interworking applies in the originating and terminating network. User A is located in Network A and user B is located in Network B. Ensure when user A, subscribed to the User-to-User service 3 essential, calls user B, not subscribed to User-to-User service 3, the call is rejected by the network. A 500 Server Internal Error is sent and an encapsulated ISUP/BICC REL is present, the Cause value is set to #29 or #69.
Configuration	User A is subscribed to the User-to-User service 3
SIP Parameter	INVITE: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM User-to-user Indicator Request service 3 essential 500 Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required REL Cause value #29 or #69
Message flow SIP (Network A)	Interconnection Interface SIP (Network B) INVITE (IAM) ➔ 🡐 500 Server Internal Error (REL) ACK ➔ Apply post test routine
Comments	Check: Is an ISUP/BICC IAM encapsulated in the initial INVITE request? Check: Is a User-to-user Indicator parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 1', 'essential'? Check: Is an ISUP/BICC REL encapsulated in the 500 response? Check: Is the Cause value set to #29 or #69 in the encapsulated REL? Repeat this test in reverse direction.

<table>
<tr><td>Test case number</td><td>SS_uus_015</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/SIP-I/UUS</td></tr>
<tr><td>Reference</td><td>5.4.3.2, 6.5, 7.1/[ITU-T Q.1912.5]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>([Network A] SE 17 AND SE 47 AND SE 63) AND ([Network B] SE 17 AND SE 47 AND SE 63)</td></tr>
<tr><td>Test purpose</td><td>SIP-I support: Indicating of User-to-User service 3 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 is, 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.</td></tr>
<tr><td>Configuration</td><td>User A is subscribed to the User-to-User service 3
User B is subscribed to the User-to-User service 3</td></tr>
<tr><td>SIP Parameter</td><td>INFO:
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
FAR
Facility indicator
user-to-user service
User-to-user Indicator
Request
service 3
not essential

INFO:
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
FAA
Facility indicator
user-to-user service
User-to-user Indicator
Response
service 3 provided

INFO
Content-Type: application/isup;version=itu-t92
Content-Disposition: signal;handling=required
USR
User-to-user Information
User Information</td></tr>
</table>

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 is the User-to-user Indicator parameter 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 is the User-to-user Indicator parameter set to 'Response', 'service 3 provided'? Check: Is an ISUP/BICC USR encapsulated in the INFO message sent from Network A to Network B containing an User-to-user Information parameter? Check: Is an ISUP/BICC USR encapsulated in the INFO message sent from Network B to Network A containing a User-to-user Information parameter? Repeat this test in reverse direction.

Test case number	SS_uus_016
Test case group	SIP-SIP/SIP-I/UUS
Reference	5.4.3.2, 6.5, 7.1/[ITU-T Q.1912.5]
SELECTION EXPRESSION	([Network A] SE 17 AND SE 47 AND SE 63) AND ([Network B] SE 17 AND SE 47 AND SE 63)
Test purpose	SIP-I support: Indicating of User-to-User service 3 during a session is established unsuccessful. BICC/ISUP – SIP-I interworking applies in the originating network. User A is located in Network A and user B is located in Network B. Ensure when user A is subscribed to the User-to-User service 3, user A is able to request the User-to-User service 3 while the session is established. The service request is rejected by Network B
Configuration	User A is subscribed to the User-to-User service 3 User B is not subscribed to the User-to-User service 3

SIP Parameter	INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FAR Facility indicator user-to-user service User-to-user Indicator Request service 3 not essential INFO: Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required FRJ Facility indicator user-to-user service User-to-user Indicator Response service 3 not provided
Message flow SIP (Network A) — Interconnection Interface — SIP (Network B) A session is already established INFO (FAR) → ← 200 OK INFO ← INFO (FRJ) 200 OK INFO → Apply post test routine	
Comments	A session is already established. Check: Is an ISUP/BICC FAR encapsulated in the INFO request sent from Network A to Network B and is the User-to-user Indicator parameter 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 is the User-to-user Indicator parameter 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 62

Test purpose	SIP-I support: Calling party subaddress can be correctly transferred in the Access Transport parameters. BICC/ISUP – SIP-I interworking applies in the originating network. User A is located in Network A and user B is located in Network B. Ensure that an ISUP/BICC ATP parameter is 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/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 62
Test purpose	SIP-I support. Called party subaddress can be correctly transferred in the Access Transport parameters. BICC/ISUP – SIP-I interworking applies in the originating network. User A is located in Network A and user B is located in Network B. Ensure that an ISUP/BICC ATP parameter is present in the encapsulated IAM of the INVITE request and contains a Called party subaddress.
Configuration	User A is subscribed to the SUB supplementary service
SIP Parameter	INVITE Content-Type: multipart/mixed;boundary=[any boundary name] --[any boundary name] Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required IAM Access transport Called party subaddress --[any boundary name]--

Message flow			
SIP (Network A)	Interconnection Interface		SIP (Network B)
	INVITE(IAM)	➔	
	Apply post test routine		

Comments	Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response? Check: Is an ISUP/BICC ATP parameter present in the encapsulated ANM containing a Called party subaddress? Repeat this test in reverse direction.

Test case number	SS_sub_003
Test case group	SIP-SIP/SIP-I/SUB
Reference	6.7/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network B] SE 17 AND SE 47 AND SE 62
Test purpose	SIP-I support. Connected party subaddress can be correctly transferred in the Access Transport parameters. BICC/ISUP – SIP-I interworking applies in the terminating network. User A is located in Network A and user B is located in Network B. Ensure that an ISUP/BICC ATP parameter is present in the encapsulated ANM of the 200 OK INVITE final response and a Connected party subaddress is contained.
Configuration	User B is subscribed to the SUB supplementary service
SIP Parameter	200 OK INVITE Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required ANM Access transport Connected party subaddress

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

Comments	Check: Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response? Check: Is an ISUP/BICC ATP parameter present in the encapsulated ANM containing a Called party subaddress? Repeat this test in reverse direction.

7.1.6.3 Terminal portability (TP)

Test case number	SS_tp_001
Test case group	SIP-SIP/SIP-I/TP
Reference	5.4.3.2/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 64
Test purpose	SIP-I support. SUS and RES messages transferred in an INFO request. BICC/ISUP – SIP-I interworking applies in the originating network. User A is located in Network A and user B is located in Network B. A session is already established. Ensure that an INFO request is sent from Network A to Network B and an ISUP SUS message is encapsulated containing a Suspend/resume indicator set to ISDN subscriber initiated. Ensure that an INFO request is sent from Network A to Network B and an ISUP RES message is encapsulated containing a Suspend/resume indicator set to ISDN subscriber initiated.
Configuration	User A is subscribed to the Terminal Portability supplementary service
SIP Parameter	INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required SUS Suspend/resume indicator ISDN subscriber initiated INFO Content-Type: application/isup;version=itu-t92 Content-Disposition: signal;handling=required RES Suspend/resume indicator ISDN subscriber initiated
Message flow SIP (Network A) Interconnection Interface SIP (Network B) A confirmed session already exists INFO(SUS) → ← 200 OK INFO INFO(RES) → ← 200 OK INFO Apply post test routine	
Comments	A session is already established Check: Is an ISUP SUS message encapsulated in the INFO request and the Suspend/resume indicator set to 'ISDN subscriber initiated'? Check: Is an ISUP RES message encapsulated in the INFO request and the Suspend/resume indicator set to 'ISDN subscriber initiated'? Repeat this test in reverse direction.

Test case number	SS_tp_002
Test case group	SIP-SIP/SIP-I/TP
Reference	5.4.3.2, 6.11.2, 6.11.2/[ITU-T Q.1912.5]
SELECTION EXPRESSION	[Network A] SE 17 AND SE 47 AND SE 64
Test purpose	SIP-I support. SUS message transferred in an INFO request call released. BICC/ISUP – SIP-I interworking applies in the originating network. User A is located in Network A and user B is located in Network B. A session is already established. Ensure that an INFO request is sent from Network A to Network B and an ISUP SUS message is encapsulated containing a Suspend/resume indicator set to ISDN subscriber initiated. Ensure that 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/NubP
Reference	5.3, 5.4/[ETSI TS 124 229]
SELECTION EXPRESSION	[Network A] SE 13
Test purpose	Request line in the INVITE contains the number portability indication. User A attempts to call user B ported to Network B. Ensure that the userinfo in the INVITE contains a destination number in the global number format, an 'rn' parameter containing the Number Portability Routing Number in a global number format with hex digits and optional the 'npdi' parameter.
Configuration	
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>[;npdi][; rn=(Number portability routing number)] @<hostname>;user = phone SIP/2.0
Message flow	SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ Apply post test routine
Comments	Check: Is the URI in the userinfo of the Request line in a global number format? Check: Is the URI rn parameter containing the Number Portability Routing Number in a global number format? Check: Is (optional) the URI parameter 'npdi' present? Check: Is the user parameter set to 'phone'? Repeat this test in reverse direction.

Test case number	SS_NP_002
Test case group	SIP-SIP/NubP
Reference	5.3, 5.4/[ETSI TS 124 229]
SELECTION EXPRESSION	NOT [Network A] SE 13
Test purpose	Request line in the INVITE without npdi parameter. The Network A does not have a Number Portability database. User A attempts to call user B ported to Network B. Ensure that the userinfo in the INVITE contains a destination number in a global number format and a npdi URI parameter is not present.
Configuration	
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>@<hostname>;user = phone SIP/2.0

Message flow			
SIP (Network A)	Interconnection Interface		SIP (Network B)
	INVITE	→	
	Apply post test routine		
Comments	Check: Is the URI in the userinfo of the Request line in a global number format without npdi parameter and number portability routing number? Check: Is the user parameter set to 'phone'? Repeat this test in reverse direction.		

7.3 Accounting

Test case number	SS_acc_001
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	Comparison of Charging Data Records > 1 sec Accounting of a confirmed session with a duration > 1 sec. 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.

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

Comments	1. Setup a call from Network A to Network B. 2. Verify whether the session confirmed. 3. Terminate the session after 5 secs. 4. Determine the duration of the session from the trace of the call monitor. 5. 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.

Test case number	SS_acc_002

Test case group	SIP-SIP/ACCOUNTING
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 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 5 secs. 4. Determine the duration of the session from the trace of the call monitor. 5. 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.

Test case number	SS_acc_003
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	Comparison of Charging Data Records > 15 mins. Accounting of a confirmed session with a duration of > 15 mins. 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 15 mins. 4. Determine the duration of the session from the trace of the call monitor. 5. 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.

Test case number	SS_acc_004
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	Comparison of Charging Data Records 25 mins. Accounting of a confirmed session with a duration of 25 mins. 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	

<table>
<tr><td colspan="2">Message flow
SIP (Network A)   Interconnection Interface   SIP (Network B)
INVITE →
← 180 Ringing
← 200 OK INVITE
ACK →
Communication
BYE →
← 200 OK BYE</td></tr>
<tr><td>Comments</td><td>1. Set up a call from Network A to Network B.
2. Verify whether the session confirmed.
3. Terminate the session after 25 mins.
4. Determine the duration of the session from the trace of the call monitor.
5. 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.</td></tr>
</table>

Test case number	SS_acc_005
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	Comparison of Charging Data Records more than 30 mins. Accounting of a confirmed session with a duration of > 30 mins. 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 35 mins. 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 • 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.

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 mins. Accounting of a confirmed session with a duration between 60 and 120 mins. 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 at the earliest 61 mins and at the latest 119 mins. 4. Determine the duration of the session from the trace of the call monitor. 5. 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.	

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	
	← 200 OK INVITE	
	ACK →	
	Communication	
	BYE →	
	← 200 OK BYE	

Comments	1. Set up a call from Network A to Network B. 2. Verify is the session confirmed. 3. Terminate the session after 24 hours. 4. Determine the duration of the session from the trace of the call monitor. 5. Compare the following information elements indicated in the 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.

Test case number	SS_acc_008
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	Comparison of Charging Data Records less than 1 sec. Accounting of a confirmed session with duration <1 sec. 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 sec. 4. Determine the duration of the session from the trace of the call monitor. 5. 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.	

Test case number	SS_acc_009
Test case group	SIP-SIP/ACCOUNTING
Reference	
SELECTION EXPRESSION	
Test purpose	Comparison of Charging Data Records session not confirmed. Accounting of an unsuccessful session in the early dialogue. Verify the duration of the call attempt stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface if applicable.
Configuration	
SIP Parameter	

Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE → ← 180 Ringing BYE/CANCEL → ← 200 OK BYE/CANCEL ← 487 Request Terminated ACK →	
Comments	1. Set up a call from Network A to Network B. 2. Verify whether an early dialogue established. 3. Terminate the early dialogue after 20 secs. 4. Determine the duration of the session from the trace of the call monitor. 5. 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.

7.4 Carrier selection

Test case number	SS_csel_001
Test case group	SIP-SIP/CS
Reference	5.7.1.10/[ETSI TS 124 229]
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15
Test purpose	User selects an operator 'call-by-call'. User A and user B are located in Network A. Ensure that user A is able to call user B and user A is able to select Network B as a selected carrier 'call-by-call'.
Configuration	User in Network A is not presubscribed
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0 INVITE: Request line sip: + <CC> <NDC> <SN>;npdi [;rn=<Number portability routing number>]@<hostname>; user=phone SIP/2.0
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE 1 → ← INVITE 2	

	Apply post test routine
Comments	Check: Is the 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 to 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_002
Test case group	SIP-SIP/CS
Reference	5.7.1.10/[ETSI TS 124 229]
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15
Test purpose	User is presubscribed to operator B. User A and user B are located in Network A. Ensure that user A is able to call user B and user A is preselected to Network B as a selected carrier.
Configuration	User in Network A is presubscribed to Network B
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0 INVITE: Request line sip: + <CC> <NDC> <SN>;npdi [;rn=<Number portability routing number>]@<hostname>; user=phone SIP/2.0
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE 1 → ← INVITE 2 Apply post test routine	
Comments	Check: Is the 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 to 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_003
Test case group	SIP-SIP/CS
Reference	5.7.1.10/[ETSI TS 124 229]
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15
Test purpose	User is presubscribed to an operator unequal to B, and overrides the preselection with call-by-call via operator B. User A and user B are located in Network A. User A is preselected to a network unequal to Network B. Ensure that user A is able to call user B and user A is able to select Network B as a selected carrier 'call-by-call'. The preselected carrier is ignored.
Configuration	User in Network A is presubscribed to Network B
SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0 INVITE: Request line sip: + <CC> <NDC> <SN>;npdi [;rn=<Number portability routing number>]@<hostname>; user=phone SIP/2.0
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE 1 → ← INVITE 2 Apply post test routine	
Comments	Check: Is the 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 to national regulations or national agreements. NOTE 2 – It is possible that further information is available in the Request line regarding the end user charging in case of Carrier selection? Repeat this test in reverse direction.

Test case number	SS_csel_004
Test case group	SIP-SIP/CS
Reference	5.7.1.10/[ETSI TS 124 229]
SELECTION EXPRESSION	[Network A] SE14 AND [Network B] SE15
Test purpose	User is presubscribed to an operator not operator B, and overrides the preselection with call-by-call via operator B. User A and user B are located in Network A. User A is preselected to a network unequal to Network B. Ensure that user A is able to call user B and user A is able to select Network B as a selected carrier 'call-by-call'. The preselected carrier is ignored.
Configuration	User in Network A is presubscribed not to Network B

SIP Parameter	INVITE: Request line sip: + <CC> <NDC> <SN>[;cic=(carrier ID)]@<hostname> user=phone SIP/2.0 INVITE: Request line sip: + <CC> <NDC> <SN>;npdi [;rn=<Number portability routing number>]@<hostname>; user=phone SIP/2.0
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE 1 → ← INVITE 2 Apply post test routine	
Comments	Check: Is the 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_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 ……. <…:cug> …… <..: cugCommunicationIndicator>11</…: cugCommunicationIndicator> <…:cug> INVITE: Request line sip: + <CC> <NDC> <SN@<hostname>;user=phone SIP/2.0 Content-Type: application/vnd.etsi.cug+xml Content-Disposition: ….;handling= required ……. <…:cug> …… <..: cugCommunicationIndicator>11</…: cugCommunicationIndicator> <…:cug>
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE 1 ➔ 🡐 INVITE 2 Apply post test routine	
Comments	Check: Is the '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: Does the XML body in the INVITE contain a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element? Check: Is the session setup not rejected?

7.5 Emergency call

Test case number	SS_ecall_001
Test case group	SIP-SIP/EmC
Reference	5.2.10, 5.7.1.14/[ETSI TS 124 229]
SELECTION EXPRESSION	
Test purpose	Request line in the INVITE. User A attempts to call a PSAP located in Network B. Ensure that the Request line in the INVITE contains the emergency number and an 'rn'

	parameter containing the PSAP routing number. In addition, location information may be present: • geolocation header • P-Access-Network-Info header • national solution to convey location information to make location information available for the PASP.
Configuration	
SIP Parameter	INVITE: Request line sip+ <(emergency number)>[; rn =+<(PASP routing number)] @hostname>;user = phone SIP/2.0
Message flow SIP (Network A) Interconnection Interface SIP (Network B) INVITE ➔ Apply post test routine	
Comments	Check: Is the URI in the userinfo of the Request line in a global number format containing the PSAP routing number? Check: Optional: Is the URI 'rn' parameter containing the PASP Routing Number? Check: Is the user parameter set to 'phone'? Check: Is the location information present in the initial INVITE request. Geolocation header PIDF-LO Element XML 'geopriv' sub element Or User-to-User header Or National solution Repeat this test in reverse direction.

7.6 Quality of service

7.6.1 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]

7.6.2 Test purposes for quality of service test (QoS)

Figure 7-1 presents the general reference configuration for the QoS test.

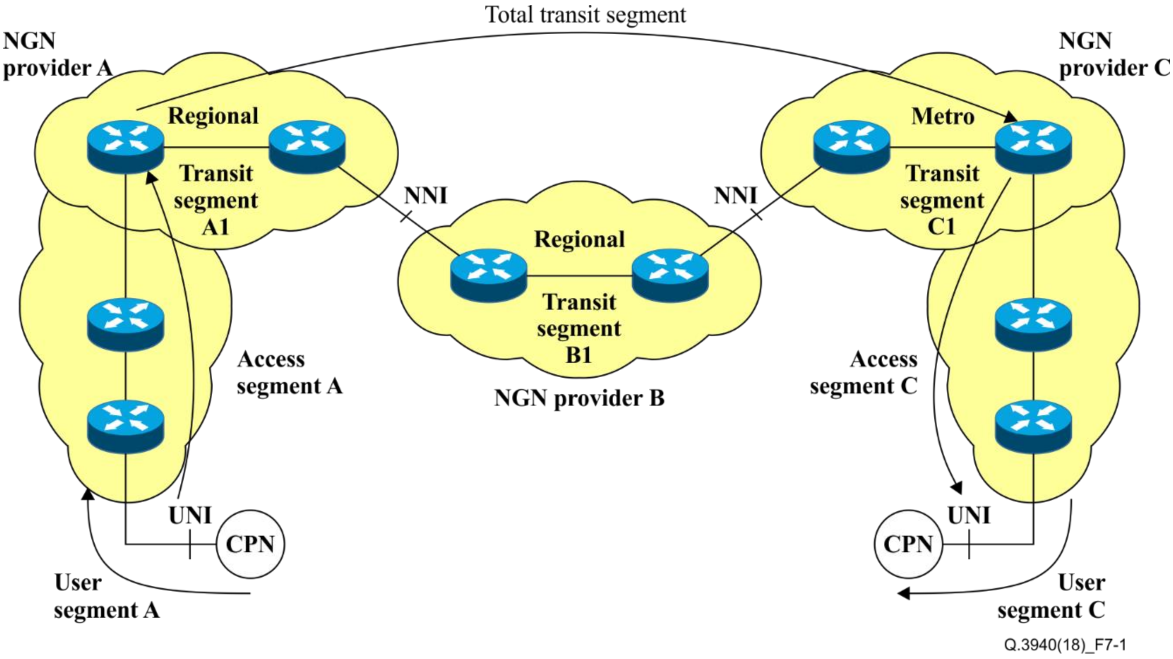

Figure 7-1 – General reference configuration

7.6.2.1 Test purposes for quality of service test (QoS)

Test case number	SS_qos_001
Test case group	SIP-SIP/QoS
Reference	
SELECTION EXPRESSION	
Test purpose	Ensure that the UE can successfully activate the voice call via dedicated voice 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; Mininmum uplink/downlink bandwith is 1 Mbit/s
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

Test case number	SS_qos_002
Test case group	SIP-SIP/QoS
Reference	[b-IETF RFC 3261]
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 $P \leq 10^{-5}$. NOTE – In Recommendation ITU-T G.826, 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}. The call is released from the calling user.
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

Test case number	SS_qos_003
Test case group	SIP-SIP/QoS
Reference	[b-IETF RFC 3261]
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 7.1-1 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.
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 7.7-1 – Call setup time (post dialling delay, PDD [b-ETSI ES 202 765-2]

<table>
<tr><th>Meaning of timers</th><th>Parameter ITU-T Q.543</th><th>IMS, PES equivalent</th><th colspan="2">Reference Load A</th><th colspan="2">Reference Load B</th></tr>
<tr><th></th><th>Detailed description</th><th></th><th>Mean Value</th><th>95% probability of not exceeding</th><th>Mean Value</th><th>95% probability of not exceeding</th></tr>
<tr><td colspan="7">VoLTE –VoLTE [b-ETSI TS 101 563]
IMS to VoLTE</td></tr>
<tr><td colspan="7">Call setup time: The definition of call setup time for VoLTE is given in [b-ETSI TS 102 250-2].</td></tr>
<tr><td></td><td></td><td></td><td>≤ 1 950 ms</td><td>≤ 2 100 ms</td><td>≤ 2 250 ms</td><td>≤ 2 400 ms
Note 1
Note 2
Note 3</td></tr>
<tr><td colspan="7">VoLTE to IMS (Note 4)</td></tr>
<tr><td colspan="7">Call setup time (post dialling delay, PDD) [b-ETSI ES 202 765-2]
To determine the call setup time in a VoIP implementation, the time in seconds from the sending of the INVITE signal through the "A" side until the receipt of the 200 OK signal is measured on the "A" side is measured, or the time in seconds from the sending of the INVITE signal through the "A" side until the receipt of the 180 Ringing signal on the "A" side is recorded.</td></tr>
<tr><td></td><td></td><td></td><td>≤ 420 ms</td><td>≤ 580 ms</td><td>≤ 750 ms</td><td>≤ 900 ms
Note 4</td></tr>
<tr><td colspan="7">IMS – IMS</td></tr>
<tr><td colspan="7">Call setup time (post dialling delay, PDD)</td></tr>
<tr><td></td><td></td><td></td><td>≤ 350 ms</td><td>≤ 500 ms</td><td>≤ 650 ms</td><td>≤ 800 ms</td></tr>
<tr><td colspan="7">NOTE 1 – Paging Cycle 1.28 s.
NOTE 2 – S1-Control plane delay: 2 ms – 15 ms (S1 is the interface between eNode Bs and MME and S-GW).
NOTE 3 – The maximum value should not exceed 5.9 seconds [b-ETSI TS 101 563].
NOTE 4 – The values are based on the condition that the originating VoLTE-UE is in ther state ECM Connected. In the case when the oLTE – UE is in ther state ECM Idle, the time duration is about 100 ms higher.</td></tr>
<tr><td colspan="7">ISDN-ISDN</td></tr>
</table>

Test case number	SS_qos_004
Test case group	SIP-SIP/QoS
Reference	[b-IETF RFC 3261]
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.
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.

Bibliography

[b-ETSI ES 202 765-2]	ETSI ES 202 765-2 V1.2.1 (2014), *Speech and multimedia Transmission Quality (STQ); QoS and network performance metrics and measurement methods; Part 2: Transmission Quality Indicator combining Voice Quality Metrics*.
[b-ETSI TS 101 563]	ETSI TS 101 563 V1.4.1 (2015), *Speech and multimedia Transmission Quality (STQ); IMS/PES/VoLTE exchange performance requirements*.
[b-ETSI TS 101 585]	ETSI TS 101 585 V1.1.2 (2012), *IMS Network Testing (INT); NGN/IMS interconnection tests at the Ic Interface; Test Suite Structure and Test Purposes (TSS&TP).*
[b-ETSI TS 102 250-2]	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*.
[b-ETSI TS 124 615]	ETSI TS 124 615 V10.2.0 (2012), *Digital cellular telecommunications system (Phase 2+);Universal Mobile Telecommunications System (UMTS); LTE; Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem*.
[b-IEC 61292-4]	IEC/TR 61292-4 ed2.0 (2010), *Optical amplifiers – Part 4: Maximum permissible optical power for the damage-free and safe use of optical amplifiers, including Raman amplifiers*.
[b-IETF RFC 3261]	IETF RFC 3261 (2002), *SIP: Session Initiation Protocol*.
[b-IETF RFC 3264]	IETF RFC 3264 (2002), *An Offer/Answer Model with Session Description Protocol (SDP).*
[b-IETF RFC 3312]	IETF RFC 3312 (2002), *Integration of Resource Management and Session Initiation Protocol (SIP).*
[b-IETF RFC 4028]	IETF RFC 4028 (2005), *Session Timers in the Session Initiation Protocol (SIP).*
[b-IETF RFC 4566]	IETF RFC 4566 (2006), *SDP: Session Description Protocol*.
[b-IETF RFC 4733]	IETF RFC 4733 (2006), *RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals*.

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