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TECHNICAL SPECIFICATION

**Core Network and Interoperability Testing (INT);**
**IMS interconnection tests at the Ic Interface;**
**(3GPP™ Release 13);**
**Test Suite Structure and Test Purposes (TSS&TP)**

Reference

RTS/INT-00134

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# Foreword

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

# Modal verbs terminology

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# 1 Scope

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

# 2 References

## 2.1 Normative references

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

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

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

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

[1] ETSI TS 129 165: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Inter-IMS Network to Network Interface (NNI) (3GPP TS 29.165 Release 13)".

[2] ETSI TS 124 229: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229 Release 13)".

[3] IETF RFC 4566 (2006): "SDP: Session Description Protocol".

[4] IETF RFC 3261 (2002): "SIP: Session Initiation Protocol".

[5] IETF RFC 3264 (2002): "An Offer/Answer Model with Session Description Protocol (SDP)".

[6] IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".

[7] ETSI TS 124 607: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Originating Identification Presentation (OIP) and Originating Identification Restriction (OIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.607 Release 13)".

[8] ETSI TS 124 608: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.608 Release 13)".

[9] ETSI TS 124 604: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Communication Diversion (CDIV) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.604 Release 13)".

[10] ETSI TS 124 605: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Conference (CONF) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.605 Release 13)".

[11] ETSI TS 124 629: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Explicit Communication Transfer (ECT) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.629 Release 13)".

[12] ETSI TS 124 611: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Anonymous Communication Rejection (ACR) and Communication Barring (CB) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.611 Release 13)".

[13] ETSI TS 124 654: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Closed User Group (CUG) using IP Multimedia (IM) Core Network (CN) subsystem, Protocol Specification (3GPP TS 24.654 Release 13)".

[14] ETSI TS 124 642: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Completion of Communications to Busy Subscriber (CCBS) and Completion of Communications by No Reply (CCNR) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.642 Release 13)".

[15] ETSI TS 124 615: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Communication Waiting (CW) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol Specification (3GPP TS 24.615 Release 13)".

[16] ETSI TS 124 606: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Message Waiting Indication (MWI) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.606 Release 13)".

[17] ETSI TS 124 610: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Communication HOLD (HOLD) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.610 Release 13)".

[18] ETSI TS 124 616: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Malicious Communication Identification (MCID) using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.616 Release 13)".

[19] ETSI TS 129 658: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; SIP Transfer of IP Multimedia Service Tariff Information; Protocol specification (3GPP TS 29.658 Release 13)".

[20] ETSI TS 124 628: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Common Basic Communication procedures using IP Multimedia (IM) Core Network (CN) subsystem; Protocol specification (3GPP TS 24.628 Release 13)".

[21] IETF RFC 5009 (September 2007): "Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media".

[22] Recommendation ITU-T V.152 (09-2010): "Procedures for supporting voice-band data over IP networks".

[23] Recommendation ITU-T T.38 (11-2015): "Procedures for real-time Group 3 facsimile communication over IP networks".

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

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

[26] IETF RFC 4733: "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals".

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

[28] Recommendation ITU-T Q.4016 (08-2016): "Testing specification of call establishment procedures based on SIP/SDP and ITU-T H.248 for a real-time fax over IP service".

[29] ETSI TS 101 563 (V1.3.1): "Speech and multimedia Transmission Quality (STQ); IMS/PES/VoLTE exchange performance requirements".

[30] ETSI ES 202 765-2 (V1.2.1): "Speech and multimedia Transmission Quality (STQ); QoS and network performance metrics and measurement methods; Part 2: Transmission Quality Indicator combining Voice Quality Metrics".

[31] Recommendation ITU-T Q.543: "Digital exchange performance design objectives".

[32] ETSI TS 102 250-2: "Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in mobile networks; Part 2: Definition of Quality of Service parameters and their computation".


## 2.2 Informative references

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

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

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


[i.1] ETSI EN 300 403-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1) protocol; Signalling network layer for circuit-mode basic call control; Part 1: Protocol specification [ITU-T Recommendation Q.931 (1993), modified]".

[i.2] ISO/IEC 9646 (1994): "Information technology -- Open Systems Interconnection -- Conformance testing methodology and framework".

[i.3] ETSI TR 102 775 (V1.5.1): "Speech and multimedia Transmission Quality (STQ); Guidance on objectives for Quality related Parameters at VoIP Segment-Connection Points; A support to NGN transmission planners".

[i.4] Recommendation ITU-T Q.1902.2 (07-2001): "Bearer Independent Call Control protocol (Capability Set 2) and Signalling System No.7 ISDN User Part: General functions of messages and parameters".

[i.5] Recommendation ITU-T G.826: "End-to-end error performance parameters and objectives for international, constant bit-rate digital paths and connections".

[i.6] ETSI TS 183 043: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation; Stage 3 specification".

[i.7] Recommendation ITUT-T Q.76: "Service procedures for Universal Personal Telecommunication - Functional modelling and information flows".

[i.8] Recommendation ITUT-T Q.764: "Signalling System No. 7 - ISDN User Part signalling procedures".

[i.9] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 Release 10)".

# 3 Definitions, conventions and abbreviations

## 3.1 Definitions

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

**Adjacent SIP Node (ASN):** SIP node (e.g. SIP Proxy or Back-to-Back User Agent or the SIP side of an IWU) that has established a direct trust relation (association) with Incoming or Outgoing IWU entities

NOTE: The SIP Proxy and Back-to-Back User Agent are defined in accordance with IETF RFC 3261 [4].

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

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

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

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

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

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

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

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

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

**test purpose:** non-formal test description, mainly using text

NOTE: TSIs test description can be used as the basis for a formal test specification (e.g. Abstract Test Suite in TTCN). See ISO/IEC 9646 [i.2].

## 3.2 Conventions for representation of SIP/SDP information

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

EXAMPLE 1: INVITE, INFO.

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

EXAMPLE 2: To, From, Call-ID.

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

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

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

EXAMPLE 4: sip: URI, tel: URI.

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

EXAMPLE 5: 100 Trying, 484 Address Incomplete.

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

EXAMPLE 6: 200 OK INVITE, 200 OK PRACK.

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

EXAMPLE 7: m=line, a=line.

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

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

## 3.3 Abbreviations

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

| | |
|---|---|
| ACR | Anonymous Communication Rejection |
| ACR-CB | Anonymous Call Rejection and Call Barring |
| AS | Application Server |
| ATP | Access Transport Parameter |
| ATS | Abstract Test Suite |
| BC | Bearer Capability |
| BCALL | Basic CALL |
| CB | Communication Barring |
| CC | Completion of Completion |
| CCBS | Completion of 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 |
| CN | Core Network |
| COLP | Connected Line Presentation |
| COLR | COnnected Line Restriction |
| CONF | Conference |
| CS | Circuit Switched |
| CUG | Closed User Group |
| CUG-OA | Closed User Group Outgoing Access |
| CW | Communication Waiting |
| DTMF | Dual Tone Multi-Frequency |
| ECT | Explicit Communication Transfer |
| FFS | For Further Study |
| GSM | Global System for Mobile Communications |
| GW | GateWay |
| HLC | High layer Compatibility |
| HOLD | Communication Hold |
| IA | In coming Access allowed |
| IBCF | Interconnection Boarder Control Function |
| ICB | Incoming Call Barring |
| IMS | IP Multimedia core network Subsystem |
| IP | Internet Protocol |
| ISDN | Integrated Services Digital Network |

| | |
|---|---|
| ISUP | ISdn User Part |
| IUT | Implementation Under Test |
| LLC | Low Layer Comaptibility |
| MCID | Malicious Communication Identification |
| MGCF | Media Gateway Control Function |
| MWI | Message Waiting Indication |
| NDUB | Network Determined User Busy |
| OA | Outgoing Access |
| OIP | Originating Identification Presentation |
| OIR | Originating Identification presentation Restriction |
| PBX | Private Branch eXchage |
| PDP | Programmable Data Processor |
| PICS | Protocol Implementation Conformance Statement |
| PIXIT | Protocol Implementation eXrtra Information |
| PLMN | Public Land Mobile Network |
| PSTN | Public Switched Telephone Network |
| PT | Payload Type |
| PTY | ParTY |
| QoS | Quality of Service |
| RTP | Realtime Transport Protocol |
| SDP | Session Description Protocol |
| SE | Selection Expression |
| SIP | Session Initiation Protocol |
| SIP-I | SIP containing ISUP |
| SS | Supplementary Sercice |
| SUB | SUB addressing |
| TIP | Terminating Identification Presentation |
| TIR | Terminating Identification Restriction |
| TP | Test Purpose |
| TSS | Test Suite Structure |
| TSS&TP | Test Suite Structure and Test Purposes |
| UA | User Agent |
| UAC | User Agent Client |
| UAS | User Agent Server |
| UDUB | User Determined User Busy |
| UE | User Equipment |
| UE-A | User Equipment A side |
| UE-B | User Equipment B side |
| URI | Uniform Resource Identifier |
| USI | User Service Information |
| UUS | User to User Service |
| XML | eXtensible Markup Language |

# 4 Test Suite Structure (TSS)

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

# 5 Declarations

## 5.1 Numbering Scheme

FFS.

## 5.2 Reference configuration

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

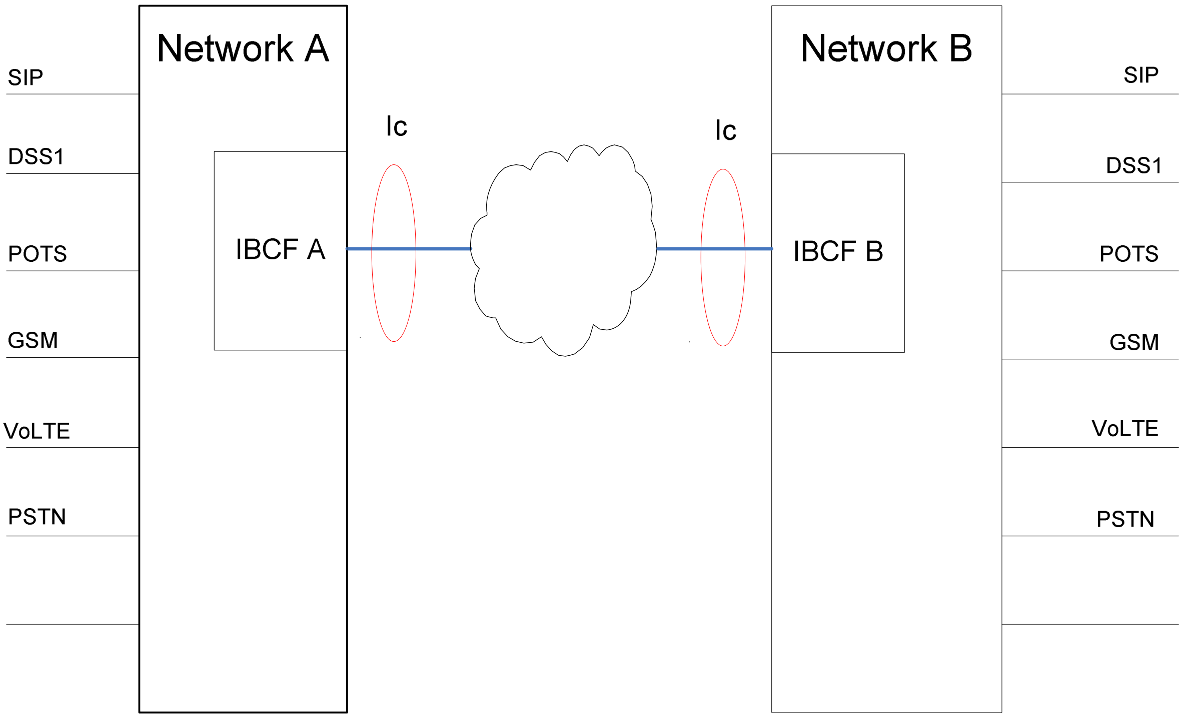


**Figure 5.2-1: Reference configuration for the interconnection test**

# 5.3 Selection of End Devices

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

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

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

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

**Table 5.3-1: Overview of end devices**

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

# 6 Selection Expressions

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

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

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

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

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

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

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

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

| SELECTION EXPRESSION: | | Support Network A | Support Network B |
|---|---|---|---|
| **SE 28:** | The Network supports Communication Forwarding Not Logged in (CFNL)? | | |
| **SE 29:** | The Network supports Communication Deflection? | | |
| **SE 30:** | The Network supports the CDIV Notification procedure? | | |
| **SE 31:** | The Network supports conference (CONF)? | | |
| **SE 32:** | The Network supports the Communication Barring procedure (CB) - (Black list for incoming calls)? | | |
| **SE 33:** | The Network supports the Anonymous Communication Rejection (ACR)? | | |
| **SE 34:** | The Network supports the Closed User Group (CUG)? | | |
| **SE 35:** | The Network supports the Communication Waiting (CW) service? | | |
| **SE 36:** | The Network supports the $T_{AS-CW}$ timer? | | |
| **SE 37:** | The Network supports Explicit Communication Transfer (ECT)? | | |
| **SE 38:** | The network supports Malicious Communication Identification (MCID)? | | |
| **SE 39:** | The Network supports Message Waiting Indication (MWI)? | | |
| **SE 40:** | The Network supports Completion of Communications to Busy Subscriber (CCBS)? | | |
| **SE 41:** | The Network supports Completion of Communications by No Reply (CCNR)? | | |
| | **Terminal capabilities** | | |
| **SE 42:** | The End device requires the resource reservation? | | |
| **SE 43:** | The End device supports Fax transmission via G.711 codec? | | |
| **SE 44:** | The End device supports Fax transmission via V.152 codec? | | |
| **SE 45:** | The End device supports Fax transmission via m-line T.38 codec? | | |
| **SE 46:** | A SIP end device is used supporting an ISDN user equipment and the PSTN XML Schema is used? | | |
| **SE 47:** | End device is located in the PSTN or PLMN? | | |
| **SE 48:** | The terminating UE supports the from-change tag procedure **and sends a second user identity in an UPDATE request after the dialogue is confirmed?** | | |
| **SE 49:** | The end device performs ECT using the 'Blind/assured transfer'? | | |
| **SE 50:** | The end device performs ECT using the '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?** | | |
| **SE 55C:** | **CDIV Mobile subscriber not reachable is supported in the PSTN/PLMN part of the network?** | | |
| **SE 55D:** | **CDIV call deflection is supported in the PSTN/PLMN part of the network?** | | |
| **SE 56:** | CONF/3PTY is supported in the PSTN/PLMN part of the network? | | |
| **SE 57:** | ACR is supported in the PSTN/PLMN part of the network? | | |
| **SE 58:** | CUG is supported in the PSTN/PLMN part of the network? | | |
| **SE 59:** | CW is supported in the PSTN/PLMN part of the network? | | |
| **SE 60:** | ECT is supported in the PSTN/PLMN part of the network? | | |
| **SE 61:** | MCID is supported in the PSTN/PLMN part of the network? | | |
| **SE 61A:** | Call Completion is supported in the PSTN/PLMN part of the network? | | |
| **SE 62:** | SUB is supported in the PSTN/PLMN part of the network? | | |
| **SE 63:** | UUS is supported in the PSTN/PLMN part of the network? | | |
| **SE 64:** | TP is supported in the PSTN/PLMN part of the network? | | |

# 7 Test purposes

## 7.0 Introduction

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

## 7.1 Testing of SIP protocol requirements

### 7.1.1 Test purposes for Basic call, Successful

| | | |
|---|---|---|
| **Test case number** | SS_bcall_001 | |
| **Test case group** | BCALL/successful | |
| **Reference** | [4] | |
| **SELECTION EXPRESSION** | | |
| **Test purpose** | **Basic call normal call clearing from the called user.**<br><br>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<br>**Check:** Ensure the property of speech.<br>**Check:** Are the media streams terminated after the 200 OK BYE was sent?<br>Repeat this test in reverse direction.<br>Repeat this test with all chosen end devices. | |

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

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

| | |
|---|---|
| **Test case number** | SS_bcall_004 |
| **Test case group** | BCALL/successful |
| **Reference** | 5.10/ [2] |
| **Testspec Reference** | |
| **SELECTION EXPRESSION** | SE 1 |
| **Test purpose** | **P-Charging-Vector header in the INVITE.**<br><br>Ensure that the P-Charging-Vector header is present in the INVITE establishes a communication between a user of network A and a user of network B and the 'icid-value' and the 'orig-ioi' parameter is present. |
| **Configuration** | |
| **SIP Parameter** | INVITE<br>P-Charging-Vector: icid-value; orig-ioi |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE ➔<br>**Apply post test routine** |
| **Comments** | Establish a communication from network A to Network B<br>**Check:** The P-Charging-Vector header contains the icid-value parameter.<br>**Check:** The P-Charging-Vector header contains the orig-ioi parameter.<br>Repeat this test in reverse direction. |

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

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

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

| | |
|---|---|
| **Test case number** | SS_bcall_008 |
| **Test case group** | BCALL/successful |
| **Reference** | 8/ [21] |
| **SELECTION EXPRESSION** | [Network A] SE 3 AND [Network B] SE 3 AND (SE 25 OR SE 26 OR SE 27 OR SE 28 OR SE 29) |
| **Test purpose** | **P-Early-Media header supported early dialogue with 181.**<br><br>Ensure that an early dialogue is established by sending a 181 Call Is Being Forwarded from Network B and the P-Early-Media header is present authorizes early media. The Call is forwarded in network B. |
| **Configuration** | **Subscription options:**<br>• Originating user receives notification that his communication has been diverted = Yes |
| **SIP Parameter** | INVITE<br>P-Early-Media: supported<br>SDP<br><br>181<br>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<br>**Check:** Is a 181 sent from Network B to establish an early dialogue?<br>**Check**: Is an SDP present in the 181 as a SDP answer?<br>**Check:** Is a bearer transmission possible in backward direction? (Optional).<br>**NOTE:** The presence of the P-Early-Media header in the INVITE request indicates the support of "early media Authorization" in the originating Network.<br>**NOTE:** The presence of the P-Early-Media header in the 181 indicates the support of the P-Early-Media header and authorizes the media in the early dialogue.<br>Repeat this test in reverse direction. |

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

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

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

| | |
|---|---|
| **Test case number** | SS_bcall_012 |
| **Test case group** | BCALL/successful |
| **Reference** | 5.10/ [2] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Record-Route header in the 180 Ringing.**<br><br>Ensure if a Record-Route header was present in the initial INVITE that the Record-Route header is present in the 180 Ringing provisional response as the first response from network B upon a connection establish setup from network A. |
| **Configuration** | |
| **SIP Parameter** | INVITE<br>Record-Route<br><br>180:<br>Record-Route |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>**Apply post test routine** | |
| **Comments** | Establish a communication from network A to Network B<br>**Check:** If the Record-Route header is present is in the 180 Ringing.<br>**NOTE:** The Record-Route header is optional.<br>Repeat this test in reverse direction.<br>Repeat this test with all chosen end devices. |

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

| | |
|---|---|
| **Test case number** | SS_bcall_014 |
| **Test case group** | BCALL/successful |
| **Reference** | 5.10/ [2] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Route header in the BYE of the terminating user.**<br><br>Ensure that if a Record-Route header was present in the initial INVITE the Route header may be present in the BYE request sent from the terminating user equipment in network B the topmost Route header or entry is set to the IBCF of network A. |
| **Configuration** | |
| **SIP Parameter** | BYE:<br>Route: <Address of IBCF in network A>;lr, …. |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>**A confirmed session already exists**<br>← BYE<br>200 OK BYE →<br>**Apply post test routine** | |
| **Comments** | Establish a communication from network A to Network B<br>**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.<br>Repeat this test in reverse direction.<br>Repeat this test with all chosen end devices. |

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

| | |
|---|---|
| **Test case number** | SS_bcall_016 |
| **Test case group** | BCALL/successful |
| **Reference** | [4] and [5] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Handling of SDP parameters in the INVITE.**<br><br>Ensure that call establishment and the correct handling of the SDP parameters of the INVITE message defined as: TYPE_SDP is performed correctly. Ensure that in the active call state the voice/data transfer on the media channels is performed correctly (e.g. testing QoS parameters). In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 7.1.1-1 applies. |
| **Configuration** | |
| **SIP Parameter** | INVITE:<br>Content-Type: application/sdp<br>m=audio <Port number> RTP/AVP TYPE_SDP= PIXIT (table 7.1.1-1)<br>*or*<br>m= Image <Port number> Udptl *or* Tcptl TYPE_SDP= PIXIT (table 7.1.1-1)<br>a=TYPE_SDP= PIXIT (table 7.1.1-1)<br>b=TYPE_SDP= PIXIT (table 7.1.1-1) |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>**Apply post test routine** |
| **Comments** | Establish a communication from network A to Network B<br>**Check:** Is the preferred codec set to TYPE_SDP?<br>**Check:** If present: Is the a line set to TYPE_SDP?<br>**Check:** If present: Is the b line set to TYPE_SDP?<br>**Check:** Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description?<br>Repeat this test in reverse direction.<br>Repeat this test with all chosen end devices. |

| | |
|---|---|
| **Test case number** | SS_bcall_017 |
| **Test case group** | BCALL/successful |
| **Reference** | [4] and [5] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **The SDP answer is sent in the 200 OK.**<br><br>Ensure that the call establishment performed correctly.<br>The initial INVITE contains an SDP with the offer 1 according table 7.1.1-1.<br>Ensure that answer related to the SDP offer is contained in the 200 OK INVITE message.<br>Ensure that in the confirmed state the voice transfer on the media and B-channels is performed correctly. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE (SDP1) →<br>← 180 Ringing<br>← 200 OK INVITE (SDP2)<br>ACK →<br>Apply post test routine |
| **Comments** | Establish a communication from network A to Network B.<br>**Check:** Is the SDP answer contained in the 200 OK INVITE?<br>**NOTE:** An SDP answer could be present in a provisional response.<br>Repeat this test in reverse direction.<br>Repeat this test with all chosen end devices. |

| | |
|---|---|
| **Test case number** | SS_bcall_018 |
| **Test case group** | BCALL/successful |
| **Reference** | [4] and [5] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **First response 200 OK INVITE.**<br><br>Ensure that call establishment and the correctly if the called user answers with a 200 OK message. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 200 OK INVITE<br>ACK →<br>**Apply post test routine** |
| Comments | Establish a communication from network A to Network B<br>**Check:** Is it possible to confirm a session without early dialogue?<br>Repeat this test in reverse direction.<br>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> (see note)** | **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 |
| 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** | [4] and [5] |
| **SELECTION EXPRESSION** | [Network A] SE 43 AND [Network B] SE 43 |
| **Test purpose** | **Fax transmission using the G.711 codec.**<br><br>Ensure that a Fax transmission is possible from Network A to Network B and the relevant codec is the G.711 codec. Ensure in the active call state the property of Fax transmission.<br>The call establishment procedures based on SIP/SDP and H.248 for a real-time fax over IP service are described in Recommendation ITU-T Q.4016 [28]. |
| **Configuration** | |
| **SIP Parameter** | INVITE: SDP<br>m=audio <Port> RTP/AVP 8/0<br>180/200 OK INVITE: SDP<br>m=audio <Port> RTP/AVP 8 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE (SDP1) →<br>← 180 Ringing<br>← 200 OK INVITE (SDP2)<br>ACK →<br>**Apply post test routine** |
| **Comments** | Establish a communication from network A to Network B<br>**Check:** Is the SDP answer contained in the 200 OK INVITE?<br>**Check:** Is Fax transmission successful?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_bcall_021 |
| **Test case group** | BCALL/successful |
| **Reference** | [5] and [22] |
| **SELECTION EXPRESSION** | [Network A] SE 44 AND [Network A] SE 44 |
| **Test purpose** | **Fax transmission using the V.152 codec.**<br><br>Ensure that a Fax transmission is possible from Network A to Network B and the relevant codec is the V.152 codec. Ensure in the active call state the property of Fax transmission.<br>The call establishment procedures based on SIP/SDP and H.248 for a real-time fax over IP service are described in Recommendation ITU-T Q.4016 [28]. |
| **Configuration** | |
| **SIP Parameter** | INVITE: SDP<br>m=audio <Port> RTP/AVP 8 <dynamic-PT><br>a=rtpmap <dynamic-PT> PCMA/8000<br>a=gpmd; vbd=yes<br><br>180/200 OK INVITE: SDP<br>m=audio <Port> RTP/AVP <dynamic-PT><br>a=rtpmap <dynamic-PT> PCMA/8000<br>a=gpmd; vbd=yes |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE (SDP1) →<br>← 180 Ringing<br>← 200 OK INVITE (SDP2)<br>ACK →<br>**Apply post test routine** |
| **Comments** | Establish a communication from network A to Network B<br>**Check:** Contains the SDP offer in the initial INVITE a voice band data codec?<br>**Check:** Contains the SDP answer in the 180 or 200 OK INVITE a voice band data codec?<br>**Check:** Is Fax transmission successful?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_bcall_022 |
| **Test case group** | BCALL/successful |
| **Reference** | [5] and [23] |
| **SELECTION EXPRESSION** | [Network A] SE 45 AND [Network B] SE 45 |
| **Test purpose** | **Fax transmission using the T.38 in an audio m-line codec.**<br><br>Ensure that a Fax transmission is possible from Network A to Network B and the relevant codec is the T.38 in an 'audio' m-line codec. Ensure in the active call state the property of Fax transmission.<br>The call establishment procedures based on SIP/SDP and H.248 for a real-time fax over IP service are described in Recommendation ITU-T Q.4016 [28]. |
| **Configuration** | |
| **SIP Parameter** | INVITE: SDP<br>m=audio <Port> RTP/AVP 8 OR <dynamic-PT><br>a=rtpmap 8 OR <dynamic-PT> PCMA/8000<br>m=image <Port> udptl t38<br><br>180/200 OK INVITE: SDP<br>m=image <Port> udptl t38 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE (SDP1) →<br>← 180 Ringing<br>← 200 OK INVITE (SDP2)<br>ACK →<br>**Apply post test routine** |
| **Comments** | Establish a communication from network A to Network B<br>**Check:** Contains the SDP offer in the initial INVITE a T.38 codec in an 'audio' line?<br>**Check:** Contains the SDP answer in the 180 or 200 OK INVITE a T.38 codec in an 'audio' line?<br>**Check:** Is Fax transmission successful?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_bcall_023 |
| **Test case group** | BCALL/successful |
| **Reference** | 4.9 and annex N/ [2] |
| **SELECTION EXPRESSION** | [Network A] SE 47 AND [Network A] SE 4 AND [Network B] SE 4 |
| **Test purpose** | **Overlap sending, the Multiple INVITE method is used.**<br><br>Ensure that call establishment using overlap sending is performed correctly.<br>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<br>**Check:** All INVITE requests contain the same Call ID and From header values.<br>SIP answers with 180 Ringing.<br>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 and annex N/ [2]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 47 AND [Network A] SE 5 AND [Network B] SE 5</td></tr>
<tr><td>Test purpose</td><td>Overlap sending, the in-Dialogue method is used<br><br>Ensure that call establishment using overlap sending is performed correctly.<br>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:<br>Supported: 100rel<br><br>183: Require: 100rel<br><br>INFO:<br>Content-Type: application/x-session-info<br>SubsequentDigit: <additional digits></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(CSq 1) 1</td><td>→</td><td></td></tr>
<tr><td></td><td>←</td><td>484 Address Incomplete(CSq 1)</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>INVITE(CSq 2) 2</td><td>→</td><td></td></tr>
<tr><td></td><td>←</td><td>183 Session Progress(CSq 2)</td><td></td><td></td></tr>
<tr><td></td><td></td><td>PRACK</td><td>→</td><td></td></tr>
<tr><td></td><td>←</td><td>200 OK PRACK</td><td></td><td></td></tr>
<tr><td></td><td></td><td>INFO</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>……..</td><td></td><td></td></tr>
<tr><td></td><td></td><td>INFO</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>180 Ringing(CSq 2)</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>Establish a communication from ISDN to SIP using the overlap operation in ISDN<br>Check: All INVITE requests contains the same Call ID and From header values.<br>Check: The 183 session Progress that establishes an early dialogue contains a Require header set to 100rel.<br>Check: All INFO requests contain the Content-Type header set to 'application/x-session-info'.<br>Check: All INFO requests contains the 'SubsequentDigit:' MIME body containing the additional digits.<br>The UE B answers with 180 Ringing response after the INVITE was received.<br>Repeat this test in reverse direction.</td></tr>
</table>

| **Test case number** | SS_bcall_025 |
|---|---|
| **Test case group** | BCALL/successful |
| **Reference** | 5.1.1.1.2/ [25] |
| **SELECTION EXPRESSION** | [Network A] (SE 46 OR SE 47) AND [Network A] SE 6 |
| **Test purpose** | **PSTN XML BearerCapability element in the INVITE**<br><br>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 [10] applies |
| **SIP Parameter** | INVITE:<br>Content-Type: application/vnd.etsi.pstn+xml<br>Content-Disposition: signal;handling=optional<br><br><?xml version="1.0" encoding="utf-8"?><br>PSTN<br>BearerCapability<br>BCoctet3<br>CodingStandard>00<<br>InformationTransferCabability>ITC_value<<br>< BCoctet4<br>TransferMode>00<<br>InformationTransferRate>10000<<br>BCoctet5<br>Layer1Identification>01<<br>UserInfoLayer1Protocol>00011< |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>**Apply post test routine** |
| **Comments** | **Check:** Is a PSTN XML MIME body contained in the INVITE request?<br>**Check:** Is the BearerCapability element present?<br>**Check:** Is the InformationTransferCabability element set as indicated in table 7.1.1-2?<br>**Check:** Is the InformationTransferCabability element value consistent with the codec list in the SDP?<br>**Check:** Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP?<br>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/ [25] |
| **SELECTION EXPRESSION** | [Network A] (SE 46 OR SE 47) AND [Network A] SE 6 |
| **Test purpose** | **PSTN XML HighLayerCapability element in the INVITE.**<br><br>User A is located in network A and an ISDN end device is used. Ensure that the INVITE request contains a PSTN XML MIME body and a HighLayerCapability element is present. |
| **Configuration** | User A is an ISDN access either in the PSTN or the SIP - ISDN interworking according [10] applies |
| **SIP Parameter** | INVITE:<br>Content-Type: application/vnd.etsi.pstn+xml<br>Content-Disposition: signal;handling=optional<br><br><?xml version="1.0" encoding="utf-8"?><br>PSTN<br>HighLayerCompatibility<br>HLOctet3<br>CodingStandard>00<<br>Interpretation>100<<br>PresentationMethod>01<<br>HLOctet4<br>HighLayerCharacteristics>[any value]< |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE ➔<br>**Apply post test routine** |
| **Comments** | **Check:** Is a PSTN XML MIME body contained in the INVITE request?<br>**Check:** Is the HighLayerCapability element present?<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_bcall_028 |
| **Test case group** | BCALL/successful |
| **Reference** | 5.1.2.2/ [25] |
| **SELECTION EXPRESSION** | [Network B] (SE 46 OR SE 47) AND [Network B] SE 6 |
| **Test purpose** | **PSTN XML ProgressIndicator element in the 180.**<br><br>User B is located in network B and an ISDN end device is used. Ensure that the 180 Ringing response contains a PSTN XML MIME body and at least one ProgressIndicator element is present. |
| **Configuration** | User B is an ISDN access either in the PSTN or the SIP - ISDN interworking according to [10] applies |
| **SIP Parameter** | 180:<br>Content-Type: application/vnd.etsi.pstn+xml<br>Content-Disposition: signal;handling=optional<br><br><?xml version="1.0" encoding="utf-8"?><br>PSTN<br>ProgressIndicator<br>ProgressOctet3<br>CodingStandard>00<<br>Location>yyyy<<br>ProgressOctet4<br>ProgressDescription>0000111<<br>*ProgressIndicator*<br>*ProgressOctet3*<br>*CodingStandard>00<*<br>*Location>0000<*<br>*ProgressOctet4*<br>*ProgressDescription>[any value]<* |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>**Apply post test routine** |
| **Comments** | **Check:** Is a PSTN XML MIME body contained in the 180 Ringing response?<br>**Check:** Is a ProgressIndicator element present and the ProgressDescription element is set to '0000111'?<br>**Check:** Is optional a second ProgressIndicator element present and the ProgressDescription element is set to any value not #2 and not #8?<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_bcall_030 |
| **Test case group** | BCALL/successful |
| **Reference** | 5.1.1.1.2/ [25] |
| **SELECTION EXPRESSION** | [Network A] (SE 46 OR SE 47) AND [Network A] SE 6 |
| **Test purpose** | **PSTN XML BearerCapability Fallback connection type element in the INVITE.**<br><br>User A is located in network A and an ISDN end device is used. Ensure that the INVITE request contains a PSTN XML MIME body and one BearerCapability element is present the InformationTransferCabability element is set to '00000' and one InformationTransferCabability element is set to '10001'. |
| **Configuration** | User A is an ISDN access either in the PSTN or the SIP - ISDN interworking according to [10] applies |
| **SIP Parameter** | INVITE:<br>Content-Type: application/vnd.etsi.pstn+xml<br>Content-Disposition: signal;handling=optional<br><br><?xml version="1.0" encoding="utf-8"?><br>PSTN<br>BearerCapability<br>BCoctet3<br>CodingStandard>00<<br>InformationTransferCabability>00000<<br>BearerCapability<br>BCoctet3<br>CodingStandard>00<<br>InformationTransferCabability>10001< |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE ➔<br>**Apply post test routine** | |
| **Comments** | **Check:** Is a PSTN XML MIME body contained in the INVITE request?<br>**Check:** Is the first BearerCapability InformationTransferCabability element set as indicated to '00000'?<br>**Check:** Is the second BearerCapability InformationTransferCabability element set as indicated to '10001'?<br>**Check:** Is the InformationTransferCabability element value consistent with the codec list in the SDP?<br>**Check:** Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP?<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_bcall_032 |
| **Test case group** | BCALL/successful |
| **Reference** | 5.1.2.3/ [25] |
| **SELECTION EXPRESSION** | [Network B] (SE 46 OR SE 47) AND [Network B] SE 6 |
| **Test purpose** | **Fall back occurs.**<br><br>User B is located in network B and an ISDN end device is used. The Fallback connection type was requested in the initial INVITE request. Ensure that the 200 OK INVITE response contains a PSTN XML MIME body and a BearerCapability element is present the InformationTransferCabability element set to '00000'. A PSTN XML MIME ProgressIndicator body is present, the ProgressDescription is set to '0000101'. |
| **Configuration** | User B is an ISDN access either in the PSTN or the SIP - ISDN interworking according to [10] applies |
| **SIP Parameter** | 200:<br>Content-Type: application/vnd.etsi.pstn+xml<br>Content-Disposition: signal;handling=optional<br><br><?xml version="1.0" encoding="utf-8"?><br>PSTN<br>BearerCapability<br>BCoctet3<br>CodingStandard>00<<br>InformationTransferCabability>00000<<br>ProgressIndicator<br>ProgressOctet4<br>ProgressDescription>0000101<<br><br>OR<br><br>No PSTN XML element |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>← 200 OK INVITE<br>ACK →<br>**Apply post test routine** | |
| **Comments** | **Check:** Is a PSTN XML MIME body contained in the 200 OK INVITE response? OR<br>If no PSTN XML MIME body contained in the 200 OK INVITE response? No further checks!<br>**Check:** Is a BearerCapability element present, the InformationTransferCabability element set to '00000'?<br>**Check:** Is a ProgressIndicator element present, the ProgressDescription is set to '0000101'?<br>**Check:** Is the InformationTransferCabability element value consistent with the codec list in the SDP?<br>**Check:** Is the InformationTransferCabability element value consistent with the bandwidth information in the SDP?<br>Repeat this test in reverse direction. |

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

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

| | |
|---|---|
| **Test case number** | SS_bcall_033 |
| **Test case group** | BCALL/successful |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 |
| **Test purpose** | **SIP-I support, Basic call, IAM present in the INVITE request.**<br><br>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.<br>Ensure that all the mandatory parameters in the IAM are present and the values are valid and the **Transmission medium requirement** parameter is consistent with the SDP. |
| **Configuration** | |
| **SIP Parameter** | INVITE:<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>**Nature of connection indicators**<br>**Forward call indicators**<br>**Calling party's category**<br>**Transmission medium requirement**<br>**Called party number**<br>*Calling party number (optional)*<br>*Optional forward call indicators (optional)*<br>*Hop counter (optional)*<br>*User service information (optional)*<br>*Access transport (optional)*<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE(IAM) →<br>← 100 Trying<br>**Apply post test routine** | |
| **Comments** | Establish a communication from network A to Network B<br>**Check:** Is an ISUP IAM encapsulated in the INVITE request?<br>**Check:** Are all the mandatory ISUP parameters present in the IAM and are the values valid?<br>**Check:** Are the values of the optional parameters in the encapsulated IAM valid?<br>**Check:** Is the 'm' line with corresponding attributes in the SDP consistent with the **Transmission medium requirement** parameter?<br>**Check:** Is the **Transmission medium requirement** value consistent with the bandwidth information in the SDP?<br>Repeat this test with all possible IAM USI and ATP combinations as indicated in table 7.1.1-3<br>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/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 4 AND SE 17 AND [Network A] SE 47 |
| **Test purpose** | **SIP-I support, Basic call, overlap signalling.**<br><br>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.<br>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<br>**Check:** Are the INVITE requests sent with the same From tag and the Call-ID?<br>**Check:** After the 180 applies, are all previous INVITE transactions are terminated with a 484 final response?<br>**Check:** Is the encapsulated IAM present in the initial INVITE request also encapsulated in any following INVITE request required for the call setup?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_bcall_035 |
| **Test case group** | BCALL/successful |
| **Reference** | 6.5/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 |
| **Test purpose** | **SIP-I support, Basic call, ACM present in the 180 response.**<br><br>Ensure that on receipt of a 180 Ringing provisional response and an SIP-I - ISUP interworking is applicable in the terminating network the Backward call indicators parameter in the encapsulated ACM is present and the values are valid.<br>Ensure that the values of the optional parameters in the encapsulated ACM are valid. |
| **Configuration** | |
| **SIP Parameter** | 180:<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ACM**<br>**Backward call indicators**<br>Called party's status indicator= **subscriber free**<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 100 Trying<br>← 180 Ringing(ACM)<br>**Apply post test routine** | |
| **Comments** | Establish a communication from network A to Network B<br>**Check:** Is an ISUP ACM message encapsulated in the 180 Ringing provisional response?<br>**Check:** Is the mandatory Backward call indicators parameter present in the encapsulated ISUP ACM and are the values valid?<br>**Check:** Are the values of optional parameters in the encapsulated ISUP ACM valid?<br>**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?<br>**Check:** Can the ringing tone be heard from the terminating side?<br><br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_bcall_036 |
| **Test case group** | BCALL/successful |
| **Reference** | 6.5/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 |
| **Test purpose** | **SIP-I support. Basic call, early ACM present in the 183 response.**<br><br>Ensure that on receipt of a 183 Session Progress provisional response and an SIP-I - ISUP interworking is applicable in the terminating network the Backward call indicators parameter in the encapsulated ACM is present and the value of the **Called party's status indicator** is set to 'no indication'.<br>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:<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ACM**<br>**Backward call indicators**<br>Called party's status indicator= **no indication**<br>**Optional backward call indicator**<br>Inband info or appropriate pattern is now available<br>**Access Transport (*optional*)**<br>Progress Indicator<br>Progress description = **Destination address is non ISDN**<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 100 Trying<br>← 183 Session Progress(ACM)<br>**Apply post test routine** |
| **Comments** | Establish a communication from network A to Network B<br>**Check:** Is an ISUP ACM message encapsulated in the 183 Session Progress provisional response?<br>**Check:** Is the mandatory Backward call indicators parameter present in the encapsulated ISUP ACM and are the values valid?<br>**Check:** Is the **Called party's status indicator** in the encapsulated ISUP ACM set to '**no indication**'?<br>**Check:** Can an early media (e.g. announcement) be heard from the terminating side?<br>**Check:** Are the values of optional parameters in the encapsulated ISUP ACM valid?<br>Repeat this test in reverse direction. |

<table>
<tr><td>Test case number</td><td>SS_bcall_037</td></tr>
<tr><td>Test case group</td><td>BCALL/successful</td></tr>
<tr><td>Reference</td><td>6.6/ [24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network B] SE 17 AND [Network B] SE 47</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. Basic call, CPG present in a 180 response.<br><br>Ensure that on receipt of a 180 Ringing provisional response and an<br>SIP-I - ISUP interworking is applicable in the terminating network the Event indicator in the encapsulated CPG is present and set to 'ALERTING'.<br>Ensure that the values of the optional parameters in the encapsulated CPG are valid.</td></tr>
<tr><td>Configuration</td><td>Select a proper destination that sends at first an early ACM and after then a CPG 'ALERTING' in the PSTN/PLMN (e.g. PBX).</td></tr>
<tr><td>SIP Parameter</td><td>180:<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>CPG<br>Event indicator = ALERTING<br><br>--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow<br>SIP (Network A) Interconnection Interface SIP (Network B)<br>INVITE →<br>← 100 Trying<br>← 183 Session Progress(ACM)<br>← 180 Ringing(CPG)<br>Apply post test routine</td></tr>
<tr><td>Comments</td><td>Establish a communication from network A to Network B<br>Check: Is an ISUP CPG message encapsulated in the 180 Ringing provisional response?<br>Check: Is the mandatory Event indicator present in the encapsulated ISUP CPG set to 'ALERTING'?<br>Check: Are the values of optional parameters in the encapsulated ISUP CPG valid?<br>Repeat this test in reverse direction.</td></tr>
</table>

<table>
<tr><td>Test case number</td><td>SS_bcall_038</td></tr>
<tr><td>Test case group</td><td>BCALL/successful</td></tr>
<tr><td>Reference</td><td>6.7/ [24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network B] SE 17 [Network B] AND SE 47</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. Basic call, ANM present in a 200 OK INVITE response.<br><br>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.<br>Ensure that the values of the optional parameters in the encapsulated ANM are valid.</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td>200 OK:<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>ANM<br><br>--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow<br>SIP (Network A) Interconnection Interface SIP (Network B)<br>INVITE →<br>← 100 Trying<br>← 180 Ringing(ACM)<br>← 200 OK INVITE(ANM)<br>ACK →<br>Apply post test routine</td></tr>
<tr><td>Comments</td><td>Establish a confirmed communication from network A to Network B<br>Check: Is an ISUP ANM encapsulated in the 200 OK INVITE?<br>Check: Are the values of optional parameters in the encapsulated ISUP ANM valid?<br>Check: Ensure the property of speech.<br>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?<br>Repeat this test in reverse direction.</td></tr>
</table>

| | |
|---|---|
| **Test case number** | SS_bcall_039 |
| **Test case group** | BCALL/successful |
| **Reference** | 5.4.3.4, 6.11.2/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 [Network A] AND SE 47 |
| **Test purpose** | **SIP-I support. Basic call, REL present in a BYE request sent from the originating network.**<br><br>Ensure that an ISUP REL message is encapsulated in a BYE request sent in the release procedure initiated from the originating user when ISUP - SIP-I interworking is applicable in the originating network.<br>Ensure the validity of the cause indicator in the encapsulated REL.<br>Ensure that the ISUP RLC is encapsulated in the 200 OK BYE. |
| **Configuration** | |
| **SIP Parameter** | BYE:<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**REL**<br>Cause value:<br><br>--[any boundary name]--<br><br><br>200 OK BYE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**RLC**<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 100 Trying<br>← 180 Ringing<br>← 200 OK INVITE<br>ACK →<br>Communication<br>BYE(REL) →<br>← 200 OK BYE(RLC) |
| **Comments** | Establish a confirmed communication from network A to Network B<br>The originating user terminates the communication<br>**Check:** Is the ISUP REL encapsulated in the BYE request?<br>**Check:** Are the cause indicators in the encapsulated ISUP REL valid?<br>**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?<br>**Check:** Is the ISUP RLC encapsulated in the 200 OK BYE?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_bcall_040 |
| **Test case group** | BCALL/successful |
| **Reference** | 5.4.3.4, 6.11.2/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 [Network B] AND SE 47 |
| **Test purpose** | **SIP-I support. Basic call, REL present in a BYE request sent from the terminating network.**<br><br>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.<br>Ensure the validity of the cause indicator in the encapsulated REL.<br>Ensure that the ISUP RLC is encapsulated in the 200 OK BYE. |
| **Configuration** | |
| **SIP Parameter** | BYE:<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**REL**<br>Cause value:<br><br>--[any boundary name]--<br><br><br>200 OK BYE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**RLC**<br><br>--[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<br>The terminating user terminates the communication<br>**Check:** Is the ISUP REL encapsulated in the BYE request?<br>**Check:** Are the cause indicators in the encapsulated ISUP REL valid?<br>**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?<br>**Check:** Is the ISUP RLC encapsulated in the 200 OK BYE?<br>Repeat this test in reverse direction. |

### 7.1.2 Codec negotiation

| | |
|---|---|
| **Test case number** | SS_codec_001 |
| **Test case group** | BCALL/Codec_Negotiation |
| **Reference** | [3], [4] and [5] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Session update requested by the calling user.**<br><br>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.<br>In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 7.1.2-1 applies. |
| **Configuration** | |
| **SIP Parameter** | SDP1: codec x chosen from table 7.1.2-1<br>SDP3: codec y chosen from table 7.1.2-1 |
| **Message flow**<br>SIP (Network A) | **Interconnection Interface** **SIP (Network B)**<br>**A confirmed session already exists (SDP 1)** |
| **CASE A** | INVITE(SDP3) →<br>← 200 OK INVITE(SDP4)<br>ACK → |
| **CASE B** | UPDATE(SDP3) →<br>← 200 OK UPDATE(SDP4)<br>**Apply post test routine** |
| **Comments** | Establish a communication from network A to Network B using SDP1 chosen from the table 7.1.2-1<br>**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.<br>**Check:** Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_codec_002 |
| **Test case group** | BCALL/Codec_Negotiation |
| **Reference** | [3], [4] and [5] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Session update requested by the called user.**<br><br>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.<br>The requestor responds to the 200 (OK) with an ACK.<br>In case when the parameter in the SDP rtpmap:<dynamic-PT> is used the codecs in table 7.1.2-1 applies. |
| **Configuration** | |
| **SIP Parameter** | SDP1: codec x chosen from table 7.1.2-1<br>SDP2: codec y chosen from table 7.1.2-1 |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>**A confirmed session already exists (SDP 1)**<br>**CASE A** INVITE(SDP3) →<br>← 200 OK INVITE(SDP4)<br>ACK →<br>**CASE B** UPDATE(SDP3) →<br>← 200 OK UPDATE(SDP4)<br>**Apply post test routine** | |
| **Comments** | Establish a connection from SIP UE 1 to SIP UE 2 using SDP1 chosen from the table 7.1.2-1<br>**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.<br>**Check:** Is the codec list consistent with the attribute(s) (bandwidth) regarding the media description?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_codec_003 |
| **Test case group** | BCALL/Codec_Negotiation |
| **Reference** | [3], [4] and [5] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **The SDP answer is contained in a 200 OK final response.**<br><br>Ensure that the call establishment performed correctly.<br>• The initial INVITE contains an SDP with the offer 1.<br>• Ensure that **answer related to the SDP offer is contained in the 200 OK INVITE** message.<br>Ensure that in the confirmed call state the voice transfer on the media channels is performed correctly. |
| **Configuration** | |
| **SIP Parameter** | INVITE: SDP offer<br><br>200: SDP answer |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE(SDP1) →<br>← 180 Ringing<br>← 200 OK INVITE(SDP2)<br>ACK →<br>**Apply post test routine** | |
| **Comments** | Establish a communication from network A to Network B<br>**Check:** Is the SDP offer contained in the initial INVITE request?<br>**Check:** Is the SDP answer contained in the 200 OK INVITE final response?<br>**NOTE:** An SDP answer could be present in a provisional response.<br>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 | | | |
| 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 | 8 000 | 1 | | |

## 7.1.3 Resource Reservation

| | |
|---|---|
| **Test case number** | SS_resource_001 |
| **Test case group** | BCALL/Resource_Reservation |
| **Reference** | [3], [4], [5] and [6] |
| **SELECTION EXPRESSION** | ([Network A] SE 7 AND [Network B] SE 7) AND ([User A] SE 42 AND [User B] SE 42) |
| **Test purpose** | **Resource reservation successful, segmented status.**<br><br>Ensure that the network is able to reserve resources for quality of service when requested from the initiating user:<br>• In the INVITE the UE requests to establish QoS preconditions for all the media streams.<br>• In the 183 Session Progress the UAS supports the QoS preconditions and requests that UAC sends a confirmation when the QoS preconditions are met.<br>• The UPDATE includes in the SDP the information about the successful QoS bidirectional mode, due to the successful bidirectional PDP context established.<br>• 200 OK UPDATE the SDP contains an indication that the UE successfully reserved the QoS in the send and receive directions. |
| **Configuration** | |

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

| | |
|---|---|
| **Test case number** | SS_resource_002 |
| **Test case group** | BCALL/Resource_Reservation |
| **Reference** | [3], [4], [5] and [6] |
| **SELECTION EXPRESSION** | (Network A] SE 7 AND ([User A] SE 42 AND NOT [User B] SE 42) |
| **Test purpose** | **Resource reservation not supported.**<br><br>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.<br>• In the INVITE the UE requests to establish QoS preconditions for all the media streams.<br>• In the 183 Session Progress: **no support** by the terminating UA is indicated.<br>Or<br>In the 180 Ringing: **no support** by the terminating UA is indicated.<br>Or<br>In the 200 OK INVITE: **no support** by the terminating UA is indicated. |
| **Configuration** | |
| **SIP Parameter** | INVITE: Supported: 100rel precondition<br>SDP1: m=audio <Port number> RTP/AVP <codec><br>a=curr:qos local none<br>a=curr:qos remote none<br>a=des:qos mandatory/optional local sendrecv<br>a=des:qos none remote sendrecv<br><br>183 Session Progress:<br>SDP2: m=audio <Port number> RTP/AVP <codec><br>Or<br>180 Ringing:<br>SDP2: m=audio <Port number> RTP/AVP <codec><br>Or<br>200 OK:<br>SDP2: m=audio <Port number> RTP/AVP <codec> |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(SDP1) →<br>**CASE A**<br>← 183 Session Progress(SDP2)<br>**CASE B**<br>← 180 Ringing(SDP2)<br>**CASE C**<br>← 180 Ringing<br>← 200 OK INVITE(SDP2)<br>ACK →<br>**Apply post test routine** |
| **Comments** | Establish a communication from network A to Network B<br>**Check:** Is the quality of service for the current state local and remote set to 'none' indicated in the SDP in the **INVITE**?<br>**Check:** Is the support of Precondition **not indicated** in the 183 Session Progress (optional)<br>**Check:** Is the support of Precondition **not indicated** in the 180 Ringing (optional)<br>**Check:** Is the support of Precondition **not indicated** in the 200 OK INVITE<br>**NOTE:** This test case is applicable with an 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** | [4] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Called number is not allocated in the assumed network.**<br><br>Ensure that, when calling to unallocated number, the network initiate call clearing to the calling user with a 404 Not Found message. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 404 Not Found<br>ACK → |
| **Comments** | Establish a communication from network A to Network B, called user number is not allocated in Network B.<br>**Check:** Is a 404 Not Found sent from Network B to Network A?<br>**Check:** In case of an interworking into the PSTN, is a Reason header cause 1 present in the 404 Not Found?<br>Repeat this test in reverse direction.<br>Repeat this test with all chosen end devices. |

| | |
|---|---|
| **Test case number** | SS_unsucc_002 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | [4] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **The network B is unable to process the request.**<br><br>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**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 503 Service unavailable<br>ACK → |
| **Comments** | Establish a communication from network A to Network B, Network B is unable to process the request.<br>**Check:** Is a 503 Service unavailable sent from Network B to Network A?<br>**Check:** In case of an interworking into the PSTN, is a Reason header present in the 503 Service unavailable?<br>Repeat this test in reverse direction.<br>Repeat this test with all chosen end devices. |

| | |
|---|---|
| **Test case number** | SS_unsucc_003 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | [4] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **The called user is network determined busy.**<br><br>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**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 486 Busy Here<br>ACK → | |
| **Comments** | Establish a communication from network A to Network B, user B is network determined user busy.<br>**Check:** Is a 486 Busy Here sent from Network B to Network A?<br>**Check:** In case of an interworking into the PSTN, is a Reason header cause 17 present in the 486 Busy Here?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_unsucc_004 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | [4] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **The called user is user determined busy.**<br><br>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**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 486 Busy Here<br>ACK → | |
| **Comments** | Establish a communication from network A to Network B, user B is user determined user busy.<br>**Check:** Is a 486 Busy Here sent from Network B to Network A?<br>**Check:** In case of an interworking into the PSTN, is a Reason header cause 17 present in the 486 Busy Here?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_unsucc_005 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | [4] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **The called user is not available on the called number.**<br><br>Ensure that when the number is changed, the network initiate call clearing to the calling user with a 410 Gone message. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 410 Gone<br>ACK → |
| **Comments** | Establish a communication from network A to Network B, user B is not allocated in Network B.<br>**Check:** Is a 410 Gone sent from Network B to Network A?<br>**Check:** In case of an interworking into the PSTN, is a Reason header present in the 486 Busy Here?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_unsucc_006 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | [4] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **The number of the called user is incomplete.**<br><br>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**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 484 Address Incomplete<br>ACK → |
| **Comments** | Establish a communication from network A to Network B, the called number is incomplete.<br>**Check:** Is a 484 Address Incomplete sent from Network B to Network A?<br>**Check:** In case of an interworking into the PSTN, is a Reason header cause 28 present in the 484 Address Incomplete?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_unsucc_007 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | [3], [4] and [5] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Session update requested by the calling user is unsuccessful, existing session remains unchanged.**<br><br>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.<br>User A in Network A attempts to change the session by sending a SDP offer to the UE in Network B.<br>Network B does not support the codec sent in the offer.<br>**Check:** Is a 488 Not Acceptable Here sent from Network B to Network A?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_unsucc_008 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | [3], [4] and [5] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Session update requested by the called user is unsuccessful, existing session remains unchanged.**<br><br>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.<br>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)**<br>INVITE →<br>← 180 Ringing<br>← 200 OK INVITE<br>ACK →<br>**Communication**<br>← INVITE<br>488 Not Acceptable Here →<br>← ACK<br>**Apply post test routine** |
| **Comments** | Establish a communication from network A to Network B.<br>User B in Network B attempts to change the session by sending a SDP offer to the UE in Network A<br>Network A does not support the codec sent in the offer.<br>**Check:** Is a 488 Not Acceptable Here sent from Network B to Network A?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_unsucc_009 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | [4] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Call clearing due to no answer from the called user initiated by the calling user.**<br><br>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** — **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>CANCEL/BYE →<br>← 200 OK CANCEL/BYE<br>← 487 Request Terminated<br>ACK → |
| **Comments** | **Check:** Is a CANCEL or BYE request is sent by the originating user?<br>**Check:** In case of an interworking into the PSTN, is a Reason header cause 16 present in the CANCEL request?<br>**Check:** Is a 487 Request Terminating sent by the terminating user?<br>**Check:** Are the media streams terminated after the 200 OK CANCEL/BYE was sent?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_unsucc_010 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | [3], [4] and [5] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Codec not supported by the called user.**<br><br>The initial INVITE contains an SDP with codecs that are not supported by the called user.<br>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**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br><br>**CASE A**<br>← 488 Not Acceptable Here<br>ACK →<br><br>**CASE B**<br>← 606 Not Acceptable<br>ACK → |
| **Comments** | Establish a call setup from network A to Network B.<br>User B in Network B does not support the codec offered in the SDP received from Network A.<br>**Check:** Is a 488 Not Acceptable Here or 606 Not Acceptable sent from Network B to Network A.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_unsucc_011 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | [4] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Call clearing due to no answer from the called user initiated by the originating network.**<br><br>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**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>→ INVITE →<br>← 180 Ringing<br>**Start timer C**<br><br>**Timeout timer C**<br>CANCEL/BYE →<br>← 200 OK CANCEL/BYE<br>← 487 Request Terminated<br>ACK → |
| **Comments** | **Check:** Is a CANCEL or BYE request sent by the originating network?<br>**Check:** In case of an interworking into the PSTN, is a Reason header cause 16 present in the CANCEL request?<br>**Check:** Is a 487 Request Terminating send from the terminating user?<br>**Check:** Are the media streams terminated after the 200 OK CANCEL/BYE was sent?<br>Repeat this test in reverse direction. |

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

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

| | |
|---|---|
| **Test case number** | SS_unsucc_013 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | 6.11.2/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 |
| **Test purpose** | **SIP-I support. The called user is busy.**<br><br>Ensure that, when the called user in the PSTN/PLMN part of Network B and ISUP - SIP-I interworking applies in Network B is busy, the network initiates call clearing to the calling user with a 486 Busy Here message. An ISUP REL message is encapsulated and the **Cause value** indicator is set to '17'. |
| **Configuration** | The called user is busy in the PSTN/PLMN part in Network B |
| **SIP Parameter** | 486:<br>Reason: Q.850;cause=17 (optional)<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>REL<br>Cause value: 17<br><br>--[any boundary name]-- |
| **Message flow**<br>SIP (Network A) | **Interconnection Interface**<br>INVITE →<br>← 486 Busy Here(REL)<br>ACK →<br>**SIP (Network B)** |
| **Comments** | Establish a communication from network A to Network B, user B in the PSTN/PLMN part of Network B is busy.<br>**Check:** Is a 486 Busy Here sent from Network B to Network A?<br>**Check:** Is an ISUP REL encapsulated and the **Cause value** indicator is set to '17'?<br>**Check:** If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_unsucc_014 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | 6.11.2/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 |
| **Test purpose** | **SIP-I support. The called user rejects the call.**<br><br>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:<br>Reason: Q.850;cause=21 (optional)<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>REL<br>Cause value: 21<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 480 Temporarily Unavailable (REL)<br>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.<br>**Check:** Is a 480 Temporarily Unavailable sent from Network B to Network A?<br>**Check:** Is an ISUP REL encapsulated and the **Cause value** indicator is set to '21'?<br>**Check:** If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_unsucc_015 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | 7.7.1/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network B] SE 47 |
| **Test purpose** | **SIP-I support. Call clearing due to no answer from the called user initiated by the calling user.**<br><br>Ensure when the early dialogue is not confirmed by the called user, the calling user located in the PSTN/PLMN part of Network A and ISUP - SIP-I interworking applies in Network A initiates call clearing to the called user with CANCEL or BYE. An ISUP REL message is encapsulated in the BYE request and the **Cause value** indicator is set to '16'. |
| **Configuration** | |
| **SIP Parameter** | 480:<br>Reason: Q.850;cause=16 (optional)<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>REL<br>Cause value: 16<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** / **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>**CASE A**<br>CANCEL →<br>← 200 OK CANCEL<br>← 487 Request Terminated<br>ACK →<br><br>**CASE B**<br>BYE(REL) →<br>← 200 OK BYE(RLC)<br>← 487 Request Terminated<br>ACK → |
| **Comments** | Establish a communication from network A to Network B, user B does not confirm the communication.<br>The originating user in the PSTN/PLMN part of Network A terminates the early dialogue.<br>**Check:** Is a CANCEL or BYE request is sent from the originating network?<br>**Check:** Is an ISUP REL encapsulated in a BYE request?<br>**Check:** Is the **Cause value** of the encapsulated REL set to '16'?<br>**Check:** If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL?<br>**Check:** Is a 487 Request Terminating send from the terminating user?<br>**Check:** Are the media streams terminated after the 200 OK CANCEL/BYE was sent?<br>**NOTE:** An ISUP REL is not encapsulated in a CANCEL request.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_unsucc_016 |
| **Test case group** | BCALL/unsuccessful |
| **Reference** | 7.7.1/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network B] SE 47 |
| **Test purpose** | **SIP-I support. Call clearing due to no answer from the called user initiated by the originating network.**<br><br>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:<br>Reason: Q.850;cause=19 (optional)<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>REL<br>Cause value: 19<br><br>--[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.<br>The ISUP timer T9 in the PSTN/PLMN expires<br>**Check:** Is a CANCEL or BYE request is sent by the originating network?<br>**Check:** Is an ISUP REL encapsulated in a BYE request?<br>**Check:** Is the **Cause value** of the encapsulated REL set to '19'?<br>**Check:** If a Reason header is present, is the cause value equal to the value in the Cause value of the encapsulated ISUP REL?<br>**Check:** Is a 487 Request Terminating send from the terminating user?<br>**Check:** Are the media streams terminated after the 200 OK CANCEL/BYE was sent?<br>**NOTE:** An ISUP REL is not encapsulated in a CANCEL request.<br>Repeat this test in reverse direction. |

## 7.1.5 Test purposes for Supplementary services

### 7.1.5.1 Test purposes for OIP

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

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

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

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

| | |
|---|---|
| **Test case number** | SS_oip_005 |
| **Test case group** | SIP-SIP/Service/OIP |
| **Reference** | 4.5.2.4/ [7] |
| **SELECTION EXPRESSION** | SE 18 AND SE 19 |
| **Test purpose** | **Special arrangement exists.**<br><br>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<br>From= original value<br>P-Asserted-Header=[any registered public user identity] |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>**Apply post test routine** |
| **Comments** | **Check:** Is the From header URI set to original value sent by the user?<br>**Check:** Is the P-Asserted-Identity set to any registered public user identity?<br>**Check:** Is the user parameter set to phone?<br>Repeat this test in reverse direction.<br>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/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND SE 47 AND SE 52 |
| **Test purpose** | **SIP-I support. ISUP Calling party number *presentation allowed* in the encapsulated IAM.**<br><br>Ensure when BICC/ISUP - SIP-I interworking applies in the originating network the BICC/ISUP IAM is encapsulated in the INVITE request. The P-Asserted-Identity header field is derived from the Calling party number in the encapsulated IAM. The 'Presentation restriction' indicator in the encapsulated IAM is set to 'allowed' no Privacy value 'id' is present in the INVITE request. |
| **Configuration** | |
| **SIP Parameter** | INVITE<br>P-Asserted-Identity=[**derived from the ISUP calling party number**]<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>**Calling party number**<br>Screening indicator<br>Network provided or user provided, verified and passed<br>Presentation restriction<br>**allowed**<br>Address signal<br><br>--[any boundary name]-- |
| **Message flow** | **SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE(IAM) ➔<br>**Apply post test routine** |
| **Comments** | **Check:** Is a BICC/ISUP IAM encapsulated in the in the INVITE request?<br>**Check:** Is the **Calling party number** present in the encapsulated IAM and the screening indicator is set to 'Network provided' or 'user provided, verified and passed' and the Presentation restriction indicator is set to 'allowed'?<br>**Check:** Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM?<br>**Check:** Is the value 'id' not present in the Privacy header field (if included)?<br>Repeat this test in reverse direction. |

<table>
<tr><td>Test case number</td><td>SS_oip_007</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/OIP</td></tr>
<tr><td>Reference</td><td>7.1.3/ [24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 17 [Network A] AND SE 47 AND SE 52 AND 52A</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. ISUP Additional Calling party number presentation allowed in the encapsulated IAM.<br><br>Ensure when BICC/ISUP - SIP-I interworking applies in the originating network the BICC/ISUP IAM is encapsulated in the INVITE request. The From field is derived from the Additional Calling party number in the encapsulated IAM. The 'Presentation restriction' indicator in the encapsulated IAM is set to 'allowed' no Privacy value 'id' is present in the INVITE request.</td></tr>
<tr><td>Configuration</td><td>The originating user in the PSTN/PLMN part of Network A is subscribed to the 'no screening option'</td></tr>
<tr><td>SIP Parameter</td><td>INVITE<br>From=[derived from the ISUP Additional calling party number]<br>P-Asserted-Identity=[derived from the ISUP calling party number]<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>IAM<br>Calling party number<br>Screening indicator<br>Network Provided<br>Presentation restriction<br>allowed<br>Address signal<br>Generic number<br>Number Qualifier Indicator<br>Additional calling party number<br>Screening indicator<br>user provided, not verified<br>Presentation restriction<br>allowed<br>Address signal<br><br>--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow<br>SIP (Network A)    Interconnection Interface    SIP (Network B)<br>INVITE(IAM) ➔<br>Apply post test routine</td></tr>
<tr><td>Comments</td><td>Check: Is a BICC/ISUP IAM encapsulated in the in the INVITE request?<br>Check: Is the Calling party number present in the encapsulated IAM and the screening indicator is set to 'Network Provided' and the Presentation restriction indicator is set to 'allowed'?<br>Check: Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM?<br>Check: Is a Generic number parameter, Number Qualifier Indicator set to Additional calling party number present and the screening indicator is set to 'user provided, not verified' and the Presentation restriction indicator is set to 'allowed'?<br>Check: Is the From header field derived from the Additional calling party number in the encapsulated IAM?<br>Check: Is the value 'id' not present in the Privacy header field (if included)?<br>Repeat this test in reverse direction.</td></tr>
</table>

### 7.1.5.2 Test purposes for OIR

| | |
|---|---|
| **Test case number** | SS_oir_001 |
| **Test case group** | SIP-SIP/Service/OIR |
| **Reference** | 4.3.2, 4.5.2.4/ [7] |
| **SELECTION EXPRESSION** | SE 20 |
| **Test purpose** | **Terminating user does not receive the identity of the originating user.**<br><br>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.<br>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<br>P-Asserted-Identity:<br>Privacy:**id** OR **header** OR **user**<br>From: <sip:anonymous@anonymous.invalid> (optional) |
| **Message flow** | **SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>**Apply post test routine** |
| **Comments** | **Check:** Is the P-Asserted-Identity is present?<br>**Check:** Is the Privacy header set to 'id' or 'header' or 'user'?<br>**Check:** Is optional the From header set to an anonymous User Identity?<br>Repeat this test in reverse direction.<br>Repeat this test with all chosen end devices. |

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

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

| | |
|---|---|
| **Test case number** | SS_oir_004 |
| **Test case group** | SIP-SIP/Service/OIR |
| **Reference** | 7.1.3/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 52 AND 52A |
| **Test purpose** | **SIP-I support. ISUP Additional Calling party number *presentation restricted* in the encapsulated IAM.**<br><br>Ensure when BICC/ISUP - SIP-I interworking applies in the originating network the BICC/ISUP IAM is encapsulated in the INVITE request. The From field is derived from the Additional Calling party number in the encapsulated IAM. The 'Presentation restriction' indicator in the Generic number parameter is set to 'allowed' no Privacy value 'id' is present in the INVITE request. |
| **Configuration** | The originating user in the PSTN/PLMN part of Network A is subscribed to the 'no screening option' |
| **SIP Parameter** | INVITE<br>P-Asserted-Identity=[**derived from the ISUP calling party number**]<br>From=[**derived from the ISUP Additional calling party number**]<br>Privacy: **id**<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>IAM<br>**Calling party number**<br>Screening indicator<br>Network Provided<br>**Presentation restriction**<br>restricted<br>**Address signal**<br>**Generic number**<br>Number Qualifier Indicator<br>**Additional calling party number**<br>Screening indicator<br>user provided, not verified<br>**Presentation restriction**<br>restricted<br>**Address signal**<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE(IAM) ➔<br>**Apply post test routine** | |
| **Comments** | **Check:** Is a BICC/ISUP IAM encapsulated in the in the INVITE request?<br>**Check:** Is the **Calling party number** present in the encapsulated IAM and the screening indicator is set to 'Network Provided' and the Presentation restriction indicator is set to 'restricted'?<br>**Check:** Is the P-Asserted-Identity header field derived from the Calling party number in the encapsulated IAM?<br>**Check:** Is a Generic number parameter, Number Qualifier Indicator set to **Additional calling party number** present and the screening indicator is set to 'user provided, not verified' and the Presentation restriction indicator is set to 'restricted'?<br>**Check:** Is the From header field derived from the Additional calling party number in the encapsulated IAM?<br>**Check:** Is the value 'id' present in the Privacy header field?<br>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/ [8] |
| **SELECTION EXPRESSION** | SE 21 |
| **Test purpose** | **Originating user receives the identity of the terminating user.**<br><br>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<br>P-Asserted-Identity: |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br><br>**CASE A** ← 180 Ringing<br><br>**CASE B** ← 183 Session Progress<br><br>**CASE C** ← 200 OK INVITE(P-Asserted-Identity)<br>**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?<br>Repeat this test in reverse direction.<br>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/ [8] |
| **SELECTION EXPRESSION** | SE 21 AND SE 22 AND [Network B] SE 48 |
| **Test purpose** | **Second identity provided in UPDATE.**<br><br>Ensure that, when the option tag "**from-change**" in the **Supported** header field is provided by the originating UE in the INVITE request and the terminating UE receives the from-change tag, the terminating user sends a 'from-change' tag in the supported header in the 200 OK INVITE a second identity is provided in the UPDATE request sent by the terminated user in the From header after the ACK was received. |
| **Configuration** | Special arrangement for the terminating user exists |
| **SIP Parameter** | INVITE<br>Supported: from-change<br><br>200 OK INVITE<br>Supported: from-change<br>P-Asserted-Identity:<br><br>UPDATE<br>From: (second user identity) |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>← 200 OK INVITE(P-Asserted-Identity)<br>ACK →<br>← UPDATE (From)<br>200 OK UPDATE →<br>**Apply post test routine** |
| **Comments** | **Check:** Is the 'from-change' tag present in the Supported header of the initial INVITE request?<br>**Check:** Is the P-Asserted-Identity present in a 180 Ringing or 183 Session Progress or in a 200 OK INVITE?<br>**Check:** Is the 'from-change' tag present in the supported header of the provisional (18x) or final (200 OK) response?<br>**Check:** Is an UPDATE request sent by the terminating user containing a From header field set to the value send by the terminating user?<br>Repeat this test in reverse direction.<br>Repeat this test with all chosen end devices. |

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

| | |
|---|---|
| **Test case number** | SS_tip_004 |
| **Test case group** | SIP-SIP/Service/TIP |
| **Reference** | 6.7/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network A] SE 47 AND SE 53 |
| **Test purpose** | **SIP-I support. The Connected number presentation allowed is present in the encapsulated 200 OK.**<br><br>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<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ANM**<br>**Connected number**<br>Screening indicator<br>Network provided or user provided, verified and passed<br>Address presentation restriction<br>**allowed**<br>Address signal<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(IAM) →<br>← 180 Ringing(ACM)<br>← 200 OK INVITE(**ANM**)<br>ACK →<br>**Apply post test routine** |
| **Comments** | **Check:** Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response?<br>**Check:** Is the Screening indicator in the encapsulated ANM set to 'Network provided' or 'user provided, verified and passed'?<br>**Check:** Is the Address presentation restriction indicator in the encapsulated ANM set to allowed?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_tip_005 |
| **Test case group** | SIP-SIP/Service/TIP |
| **Reference** | 6.7/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network A] SE 47 AND SE 53 AND 53A |
| **Test purpose** | **SIP-I support. The additional connected number restricted is present in the encapsulated 200 OK.**<br><br>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'.<br>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<br>P-Asserted-Identity=[**derived from the ISUP Connected number**]<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ANM**<br>**Connected number**<br>Screening indicator<br>Network provided or user provided, verified and passed<br>Presentation restriction<br>**allowed**<br>Address signal<br>**Generic number**<br>Number Qualifier Indicator<br>**Additional connected number**<br>Screening indicator<br>user provided, not verified<br>Address Presentation Restricted<br>allowed<br>Address signal<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(IAM) →<br>← 180 Ringing(ACM)<br>← 200 OK INVITE(**ANM**)<br>ACK →<br>**Apply post test routine** |
| **Comments** | **Check:** Is the BICC/ISUP ANM encapsulated in the 200 OK INVITE final response?<br>**Check:** Is a Generic number parameter present in the encapsulated ANM?<br>**Check:** Is the Number Qualifier Indicator of the Generic number set to 'additional connected number'?<br>**Check:** Is the Screening indicator of the Generic number set to 'user provided, not verified'?<br>**Check:** Is the Address presentation restriction indicator in the Generic number set to 'allowed'?<br>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/ [8] |
| **SELECTION EXPRESSION** | SE 23 |
| **Test purpose** | **Originating user does not receive the identity of the terminating user.**<br><br>Ensure that, when the preconditions are fulfilled to prevent the presentation of the terminating user identity at the originating user,<br>the originating UE receives, in any non-100 SIP response (e.g. 180, 183, 200), a Privacy header field is set to "id" and no P-Asserted-Identity header field is present. |
| **Configuration** | The terminating user subscribes to the 'TIR' service |
| **SIP Parameter** | 18x/200 OK INVITE<br>P-Asserted-Identity:<br>Privacy: id |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>**CASE A** ← 180 Ringing<br>**CASE B** ← 183 Session Progress<br>**CASE C** ← 200 OK INVITE(P-Asserted-Identity)<br>**Apply post test routine** |
| **Comments** | **Check:** Is the P-Asserted-Identity is present in the provisional (18x) or final (200 OK) response?<br>**Check:** Is the Privacy header in the provisional (18x) or final (200 OK) response is set to 'id'?<br>Repeat this test in reverse direction.<br>Repeat this test with all chosen end devices. |

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

| | |
|---|---|
| **Test case number** | SS_tir_002 |
| **Test case group** | SIP-SIP/Service/TIR |
| **Reference** | 6.7/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 53 |
| **Test purpose** | **SIP-I support. The Connected number presentation allowed is present in the encapsulated 200 OK.**<br><br>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<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ANM**<br>**Connected number**<br>Screening indicator<br>Network provided or user provided, verified and passed<br>Address presentation restriction<br>restricted<br>Address signal<br><br>--[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?<br>**Check:** Is the Screening indicator in the encapsulated ANM set to 'Network provided' or 'user provided, verified and passed'?<br>**Check:** Is the Address presentation restriction indicator in the encapsulated ANM set to allowed?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_tir_003 |
| **Test case group** | SIP-SIP/Service/TIR |
| **Reference** | 6.7/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] 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.**<br><br>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'.<br>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<br>P-Asserted-Identity=[**derived from the ISUP Connected number**]<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ANM**<br>**Connected number**<br>Screening indicator<br>Network provided or user provided, verified and passed<br>Presentation restriction<br>restricted<br>Address signal<br>**Generic number**<br>Number Qualifier Indicator<br>**Additional connected number**<br>Screening indicator<br>user provided, not verified<br>Address Presentation Restricted<br>restricted<br>Address signal<br><br>--[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?<br>**Check:** Is a Generic number parameter present in the encapsulated ANM?<br>**Check:** Is the Number Qualifier Indicator of the Generic number set to 'additional connected number'?<br>**Check:** Is the Screening indicator of the Generic number set to 'user provided, not verified'?<br>**Check:** Is the Address presentation restriction indicator in the Generic number set to 'allowed'?<br>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/ [17] |
| **SELECTION EXPRESSION** | SE 24 |
| **Test purpose** | **Hold the session the media stream was previously set to sendrecv.**<br><br>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**<br>**SIP (Network A)**<br><br>**CASE A**<br><br><br><br>**CASE B** | **Interconnection Interface** **SIP (Network B)**<br>**A confirmed session already exists**<br>INVITE(**sendonly**) →<br>← 200 OK INVITE (recvonly)<br>ACK →<br><br>UPDATE(**sendonly**) →<br>← 200 OK UPDATE (**recvonly**)<br>**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?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_hold_002 |
| **Test case group** | SIP-SIP/Service/HOLD |
| **Reference** | 4.5.2.1/ [17] |
| **SELECTION EXPRESSION** | SE 24 |
| **Test purpose** | **Hold the session the media stream was previously set to recvonly.**<br><br>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".<br>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 the version parameter in the SDP 'o' line is incremented?<br>**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?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_hold_003 |
| **Test case group** | SIP-SIP/Service/HOLD |
| **Reference** | 4.5.2.1/ [17] |
| **SELECTION EXPRESSION** | SE 24 |
| **Test purpose** | **Resume the session the media stream was previously set to sendonly.**<br><br>Ensure that the UE A is requested to resume the session with user B the UE-A starts sending media and sends an INVITE or UPDATE request to resume the session with the attribute "a=sendrecv in the SDP. The UE A after requesting to resume the held session *receives* 200 OK final response and optionally the attribute "a=sendrecv in the SDP. The a=sendrecv attribute is the default value therefore the attribute can be omitted. |
| **Configuration** | |
| **SIP Parameter** | |

| | | | |
|---|---|---|---|
| **Message flow** | | | |
| **SIP (Network A)** | | **Interconnection Interface** | **SIP (Network B)** |
| | | **A 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 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?<br>**Check:** Is the user in network A able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? The absence of the 'sendrecv' attribute is the default value.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_hold_004 |
| **Test case group** | SIP-SIP/Service/HOLD |
| **Reference** | 4.5.2.1/ [17] |
| **SELECTION EXPRESSION** | SE 24 |
| **Test purpose** | **Resume the session the media stream was previously set to inactive.**<br><br>The Session is in the "inactive" state. Ensure that the UE A is requesting to resume the session with user B the UE-A sends an INVITE or UPDATE to resume the session with the attribute "a=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 (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 the version parameter in the SDP 'o' line is incremented?<br>**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?<br>**Check:** Is the user in network A able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_hold_005 |
| **Test case group** | SIP-SIP/Service/HOLD |
| **Reference** | 4.5.2.1/ [17] |
| **SELECTION EXPRESSION** | SE 24 |
| **Test purpose** | **Hold the session the media stream was previously set to sendrecv.**<br><br>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 the version parameter in the SDP 'o' line is incremented?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_hold_006 |
| **Test case group** | SIP-SIP/Service/HOLD |
| **Reference** | 4.5.2.1/ [17] |
| **SELECTION EXPRESSION** | SE 24 |
| **Test purpose** | **Hold the session the media stream was previously set to sendonly.**<br><br>The Session is in the held state done by UE-A. Ensure that the UE B sends an INVITE or UPDATE request to hold the session. Hold is done containing the SDP with the attribute "a=inactive". The UE A after receiving the hold request *sends* 200 OK final response containing the SDP with the attribute "a=inactive" 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?<br>**Check:** Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_hold_007 |
| **Test case group** | SIP-SIP/Service/HOLD |
| **Reference** | 4.5.2.1/ [17] |
| **SELECTION EXPRESSION** | SE 24 |
| **Test purpose** | **Resume the session the media stream was previously set to recvonly.**<br><br>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 the version parameter in the SDP 'o' line is incremented?<br>**Check:** Is the user in network B able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?<br>Repeat this test in reverse direction. |

<table>
<tr><td>Test case number</td><td>SS_hold_008</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/HOLD</td></tr>
<tr><td>Reference</td><td>4.5.2.1/ [17]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>SE 24</td></tr>
<tr><td>Test purpose</td><td>Resume the session the media stream was previously set to inactive.<br><br>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 A, 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".</td></tr>
<tr><td>Configuration</td><td></td></tr>
<tr><td>SIP Parameter</td><td></td></tr>
<tr><td colspan="2">Message flow<br>SIP (Network A) | Interconnection Interface | SIP (Network B)<br>A confirmed session already exists<br>CASE A<br>INVITE (sendonly) →<br>← 200 OK INVITE (recvonly)<br>ACK →<br>← INVITE (inactive)<br>200 OK INVITE (inactive) →<br>← ACK<br>← INVITE (recvonly)<br>200 OK INVITE (sendonly) →<br>← ACK<br><br>CASE B<br>INVITE (sendonly) →<br>← 200 OK INVITE (recvonly)<br>ACK →<br>← UPDATE (inactive)<br>200 OK UPDATE (inactive) →<br>← UPDATE (recvonly)<br>200 OK UPDATE (sendonly) →<br><br>CASE C<br>UPDATE (sendonly) →<br>← 200 OK UPDATE (recvonly)<br>← INVITE (inactive)<br>200 OK INVITE (inactive) →<br>← ACK<br>← INVITE (recvonly)<br>200 OK INVITE (sendonly) →<br>← ACK<br><br>CASE D<br>UPDATE (sendonly) →<br>← 200 OK UPDATE (recvonly)<br>← UPDATE (inactive)<br>200 OK UPDATE (inactive) →<br>← UPDATE (recvonly)<br>200 OK UPDATE (sendonly) →<br>Apply post test routine</td></tr>
<tr><td>Comments</td><td>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?<br>Check: Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?<br>Check: Is the user in network B able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?<br>Repeat this test in reverse direction.</td></tr>
</table>

| | |
|---|---|
| **Test case number** | SS_hold_009 |
| **Test case group** | SIP-SIP/Service/HOLD |
| **Reference** | 4.5.2.1/ [17] |
| **SELECTION EXPRESSION** | SE 24 |
| **Test purpose** | **Resume the session on both sides the media stream was previously set to inactive.**<br><br>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.<br>The UE A after requests to resume the session *receives* 200 OK final response containing the SDP with the attribute "a=**sendonly**.<br>The UE B after requests to resume the session sends an INVITE or UPDATE request 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 the version parameter in the SDP 'o' line is incremented?<br>**Check:** Is the user in network B able to set the session on hold by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?<br>**Check:** Is the user in network A able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?<br>**Check:** Is the user in network B able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? The absence of the 'sendrecv' attribute is the default value.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_hold_010 |
| **Test case group** | SIP-SIP/Service/HOLD |
| **Reference** | 4.5.2.1/ [17] |
| **SELECTION EXPRESSION** | SE 24 |
| **Test purpose** | **Resume the session on both sides the media stream was previously set to inactive.**<br><br>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**".<br>The UE A after receiving the Resume of the session *sends* 200 OK final response containing the SDP with the attribute "a=**sendonly**".<br>The UE A after requests to resume the session *sends an INVITE or UPDATE request* containing the SDP with the attribute "a=**sendrecv**.<br>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**<br>**A confirmed session already exists** | | **SIP (Network B)** |
|---|---|---|---|---|
| **CASE A** | ← | INVITE(sendonly) | | |
| | | 200 OK INVITE (recvonly) | → | |
| | ← | ACK | | |
| | | INVITE(inactive) | → | |
| | ← | 200 OK INVITE (inactive) | | |
| | | ACK | → | |
| | ← | INVITE(**recvonly**) | | |
| | | 200 OK INVITE (sendonly) | → | |
| | ← | ACK | | |
| | | INVITE(**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) | → | |
| | | 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 the version parameter in the SDP 'o' line is incremented?<br>**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?<br>**Check:** Is the user in network B able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented?<br>**Check:** Is the user in network A able to retrieve the session by sending an INVITE or UPDATE request and the version parameter in the SDP 'o' line is incremented? The absence of the 'sendrecv' attribute is the default value.<br>Repeat this test in reverse direction. |

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

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

<table>
<tr><td>Test case number</td><td>SS_hold_013</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/HOLD</td></tr>
<tr><td>Reference</td><td>B.10/ [24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 17 AND [Network A] SE 47 AND SE 54</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. Hold requested by the originating user, Hold by the terminating user. Retrieve requested by the originating user.<br><br>Ensure the hold and retrieve procedure when ISUP - SIP-I interworking applies in the network A:<br>• Originating user in network A places the session on hold.<br>• Terminating user in network B places the session on hold.<br>• Originating user in network A retrieves the session.<br>• Terminating user in network B retrieves the session.<br>Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the network A.</td></tr>
<tr><td>SIP Parameter</td><td>INVITE:<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>CPG<br>Generic notification<br>remote hold<br>or<br>remote retrieval<br>--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow<br>SIP (Network A) Interconnection Interface SIP (Network B)<br>A confirmed session already exists<br>INVITE(sendonly, CPG hold) →<br>← 200 OK INVITE (recvonly)<br>ACK →<br><br>← INVITE(inactive)<br>200 OK INVITE (inactive) →<br>← ACK<br><br>INVITE(sendonly, CPG retrieval) →<br>← 200 OK INVITE (recvonly)<br>ACK →<br><br>← INVITE(sendrecv)<br>200 OK INVITE (sendrecv) →<br>← ACK<br>Apply post test routine</td></tr>
<tr><td>Comments</td><td>Establish a session from network A to network B<br>The user in the PSTN/PLMN part of Network A places the session on hold.<br>Check: Is a CPG encapsulated in the INVITE request?<br>Check: Is a Generic notification parameter present the Notification indicator set to 'remote hold'?<br>Check: Is the 'a' attribute in the SDP set to 'sendonly'?<br>Check: Is the Version parameter in the SDP incremented?<br>The user in Network B places the session on hold<br>Check: Is the 'a' attribute in the SDP set to 'inactive'?<br>Check: Is the Version parameter in the SDP incremented?<br>The user in Network A retrieves the session<br>Check: Is a CPG encapsulated in the INVITE request?<br>Check: Is a Generic notification parameter present the Notification indicator set to 'remote retrieval'?<br>Check: Is the 'a' attribute in the SDP set to 'sendonly'?<br>Check: Is the Version parameter in the SDP incremented?<br>The user in Network B retrieves the session<br>Check: Is the 'a' attribute in the SDP set to 'sendrecv'?<br>Check: Is the Version parameter in the SDP incremented?<br>Repeat this test in reverse direction.</td></tr>
</table>

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

| | |
|---|---|
| **Test case number** | SS_hold_015 |
| **Test case group** | SIP-SIP/Service/HOLD |
| **Reference** | B.10/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 54 |
| **Test purpose** | **SIP-I support. Hold requested by the terminating user, Hold by the originating user. Retrieve requested by the originating user.**<br><br>Ensure the hold and retrieve procedure when ISUP - SIP-I interworking applies in the Network A:<br>• Terminating user in Network B places the session on hold.<br>• Originating user in Network A places the session on hold.<br>• Originating user in Network A retrieves the session.<br>• Terminating user in Network B retrieves the session.<br>Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A. |
| **SIP Parameter** | INVITE:<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**CPG**<br>Generic notification<br>remote hold<br>or<br>remote retrieval<br>--[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<br>The user in Network B places the session on hold.<br>**Check:** Is the 'a' attribute in the SDP set to '**sendonly**'?<br>**Check:** Is the Version parameter in the SDP incremented?<br>The user in Network A places the session on hold<br>**Check:** Is a CPG encapsulated in the INVITE request?<br>**Check:** Is a Generic notification parameter present the Notification indicator set to '**remote hold**'?<br>**Check:** Is the 'a' attribute in the SDP set to '**inactive**'?<br>**Check:** Is the Version parameter in the SDP incremented?<br>The user in Network A retrieves the session<br>**Check:** Is a CPG encapsulated in the INVITE request?<br>**Check:** Is a Generic notification parameter present the Notification indicator set to '**remote retrieval**'?<br>**Check:** Is the 'a' attribute in the SDP set to '**recvonly**'?<br>**Check:** Is the Version parameter in the SDP incremented?<br>The user in Network B retrieves the session<br>**Check:** Is the 'a' attribute in the SDP set to '**sendrecv**'?<br>**Check:** Is the Version parameter in the SDP incremented?<br>Repeat this test in reverse direction. |

<table>
<tr><td>Test case number</td><td>SS_hold_016</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/HOLD</td></tr>
<tr><td>Reference</td><td>B.10/ [24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network A] SE 17 AND [Network A] SE 47 AND SE 54</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. Hold requested by the terminating user, Hold by the originating user. Retrieve requested by the terminating user.<br><br>Ensure the hold and retrieve procedure when ISUP - SIP-I interworking applies in the Network A:<br>• Terminating user in Network B places the session on hold.<br>• Originating user in Network A places the session on hold.<br>• Terminating user in Network B retrieves the session.<br>• Originating user in Network A retrieves the session.<br>Verify the Generic notification parameter in the encapsulated CPG present in the INVITE request from the Network A.</td></tr>
<tr><td>SIP Parameter</td><td>INVITE:<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>CPG<br>Generic notification<br>remote hold<br>or<br>remote retrieval<br>--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow<br>SIP (Network A) | Interconnection Interface | SIP (Network B)<br>A confirmed session already exists<br>← INVITE(sendonly)<br>200 OK INVITE (recvonly) →<br>← ACK<br><br>INVITE(inactive, CPG hold) →<br>← 200 OK INVITE (inactive)<br>ACK →<br><br>← INVITE(recvonly)<br>200 OK INVITE (sendonly) →<br>← ACK<br><br>INVITE(sendrecv, CPG retrieval) →<br>← 200 OK INVITE (sendrecv)<br>ACK →<br>Apply post test routine</td></tr>
<tr><td>Comments</td><td>Establish a session from Network A to Network B<br>The user in Network B places the session on hold.<br>Check: Is the 'a' attribute in the SDP set to 'sendonly'?<br>Check: Is the Version parameter in the SDP incremented?<br>The user in Network A places the session on hold<br>Check: Is a CPG encapsulated in the INVITE request?<br>Check: Is a Generic notification parameter present the Notification indicator set to 'remote hold'?<br>Check: Is the 'a' attribute in the SDP set to 'inactive'?<br>Check: Is the Version parameter in the SDP incremented?<br>The user in Network B retrieves the session<br>Check: Is the 'a' attribute in the SDP set to 'sendonly'?<br>Check: Is the Version parameter in the SDP incremented?<br>The user in Network A retrieves the session<br>Check: Is a CPG encapsulated in the INVITE request?<br>Check: Is a Generic notification parameter present the Notification indicator set to 'remote retrieval'?<br>Check: Is the 'a' attribute in the SDP set to 'sendrecv'?<br>Check: Is the Version parameter in the SDP incremented?<br>Repeat this test in reverse direction.</td></tr>
</table>

### 7.1.5.6 Communication Diversion (CDIV)

#### 7.1.5.6.1 Communication Forwarding Unconditional (CFU)

| | |
|---|---|
| **Test case number** | SS_cfu_001 |
| **Test case group** | SIP-SIP/Service/CFU |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 25 |
| **Test purpose** | **Communication forwarding unconditional, basic rules.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFU.<br>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**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFU is performed**<br>← INVITE(Call-ID B-C)<br>180 Ringing(Call-ID C-B) →<br>← 180 Ringing(Call-ID B-A)<br>200 OK INVITE(Call-ID C-B) →<br>← ACK(Call-ID B-C)<br>← 200 OK INVITE(Call-ID B-A)<br>ACK(Call-ID A-B) →<br>**Communication**<br>**Apply post test routine** |
| **Comments** | **Check:** CDIV unconditional is successful.<br>**Check:** In the active call state, ensure the property of speech.<br>**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?<br>Repeat this test in reverse direction. |

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

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

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

| | |
|---|---|
| **Test case number** | SS_cfu_005 |
| **Test case group** | SIP-SIP/Service/CFU |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 25 AND SE 30 |
| **Test purpose** | **Communication forwarding unconditional, diverted-to user does not receive the URI of the served user.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFU "Served user allows the presentation of his/her URI to the diverted-to user"= No.<br>Ensure that when user A calls user B, the call is forwarded unconditional to user C, user C is not informed of the forwarding number. |
| **Configuration** | **Subscription options:**<br>• Served user allows the presentation of his/her URI to the diverted-to user = No |
| **SIP Parameter** | INVITE:<br>History-Info:<br><sip:userB@networkB?Privacy=history>;index=1,<br><sip: userC@networkA;cause=302>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFU is performed**<br>← INVITE(Call-ID B-C)<br>**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'.<br>**Check:** Is the cause parameter in the last entry is set to '302'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfu_006 |
| **Test case group** | SIP-SIP/Service/CFU |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 25 AND SE 30 |
| **Test purpose** | **Communication forwarding unconditional, diverted-to user receives the URI of the served user.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFU "Served user allows the presentation of his/her URI to diverted-to user" = Yes.<br>Ensure that when user A calls user B, the call is forwarded unconditional to user C, user C is informed of the forwarding number. |
| **Configuration** | **Subscription options:**<br>• Served user allows the presentation of his/her URI to diverted-to user = Yes |
| **SIP Parameter** | INVITE:<br>History-Info<br><sip:userB@networkB>;index=1,<br><sip: userC@networkA;cause=302>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFU is performed**<br>← INVITE(Call-ID B-C)<br>**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.<br>**Check:** Is the cause parameter in the last entry is set to '302'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfu_007 |
| **Test case group** | SIP-SIP/Service/CFU |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 25 AND SE 30 |
| **Test purpose** | **Communication forwarding unconditional, full notification.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFU Originating user receives notification that his communication has been diverted = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, and "Served user allows the presentation of his/her URI to diverted-to user" = Yes.<br>Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number. |
| **Configuration** | **Subscription options:**<br>• Originating user receives notification that his communication has been diverted = Yes<br>• Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes<br>• Served user allows the presentation of his/her URI to diverted-to user = Yes |
| **SIP Parameter** | INVITE:<br>History-Info:<br><sip:userB@networkB>;index=1,<br><sip: userC@networkA;cause=302>;index=1.1<br><br>181 Being Forwarded<br>P-Asserted-Identity: <userB@NetworkB><br>History-Info<br><sip:userB@networkB>;index=1,<br><sip: userC@networkA;cause=302>;index=1.1<br><br>200 OK INVITE<br>History-Info header:<br><sip:userB@networkB>;index=1,<br><sip: userC@networkA;cause=302>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFU is performed**<br>← INVITE(Call-ID B-C)<br>← 181 Being Forwarded(Call-ID B-A<br>180 Ringing(Call-ID C-B) →<br>← 180 Ringing(Call-ID B-A)<br>200 OK INVITE(Call-ID C-B) →<br>← ACK(Call-ID C-B)<br>← 200 OK INVITE(Call-ID B-A)<br>ACK(Call-ID A-B) →<br>**Communication**<br>**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.<br>**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.<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfu_008 |
| **Test case group** | SIP-SIP/Service/CFU |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 25 |
| **Test purpose** | **Communication forwarding unconditional, unsuccessful UDUB.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFU.<br>Ensure that when user A calls user B, the call is forwarded unconditional to user C user C is user determined user busy. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow** | |

| **SIP (Network A)** | | **Interconnection Interface** | | **SIP (Network B)** |
|---|---|---|---|---|
| | | INVITE(Call-ID A-B) | → | |
| | | **CFU is performed** | | |
| | ← | INVITE(Call-ID B-C) | | |
| | | 486 Busy Here(Call-ID C-B) | → | |
| | ← | ACK(Call-ID B-C) | | |
| | ← | 486 Busy Here(Call-ID A-B) | | |
| | | ACK(Call-ID A-B) | → | |

| | |
|---|---|
| **Comments** | **Check:** The dialogue is terminated by receiving a 486 Busy Here.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfu_009 |
| **Test case group** | SIP-SIP/Service/CFU |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 25 |
| **Test purpose** | **Communication forwarding unconditional, unsuccessful NDUB.**<br><br>The user A and user C are in network A. The user B is in network B.<br>Ensure that when user A calls user B, the call is forwarded unconditional to user C and user C is network determined user busy. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow** | |

| **SIP (Network A)** | | **Interconnection Interface** | | **SIP (Network B)** |
|---|---|---|---|---|
| | | INVITE(Call-ID A-B) | → | |
| | | **CFU is performed** | | |
| | ← | INVITE(Call-ID B-C) | | |
| | | 486 Busy Here(Call-ID C-B) | → | |
| | ← | ACK(Call-ID B-C) | | |
| | ← | 486 Busy Here(Call-ID A-B) | | |
| | | ACK(Call-ID A-B) | → | |

| | |
|---|---|
| **Comments** | **Check:** The dialogue is terminated by receiving a 486 Busy Here.<br>Repeat this test in reverse direction. |

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

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

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

| | |
|---|---|
| **Test case number** | SS_cfu_013 |
| **Test case group** | SIP-SIP/Service/CFU |
| **Reference** | 6.5/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55 |
| **Test purpose** | **SIP-I support. CFU performed in Network B, Notification subscription options is set to presentation allowed with redirection number.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number.<br>Ensure that when user A calls user B, the call is forwarded unconditional to user C, user A is notified of call diversion and informed of the diverted-to number.<br>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:**<br>• Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number |
| **SIP Parameter** | 183 Session Progress<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ACM**<br>Backward call indicator<br>Called party's status indicator<br>no indication<br>Redirection number<br>Address signal (*Diverted-to user*)<br>Call diversion information<br>Notification subscription options<br>presentation allowed with redirection number<br>Redirecting reason<br>unconditional<br>Generic notification<br>call is diverting<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFU is performed**<br>← INVITE(Call-ID B-C, IAM)<br>← 183 Session Progress (Call-ID B-A, ACM)<br>**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.<br>**Check:** 183 Session Progress is received at the interconnection interface.<br>**Check:** Is an ACM encapsulated in the 183?<br>**Check:** Is the Called party's status indicator set to 'no indication'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Notification subscription options indicator set to 'presentation allowed with redirection number'?<br>**Check** Is the Redirecting reason set to 'unconditional'?<br>Repeat this test in reverse direction. |

<table>
<tr><td>Test case number</td><td>SS_cfu_014</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CFU</td></tr>
<tr><td>Reference</td><td>6.7/ [24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network B] SE 17 AND [Network B] SE 47 AND SE 53</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. CFU performed in Network B, Restriction of the Redirection number.<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Diverted-to user is subscribed to the COLR service in Permanent mode.<br>Ensure that when user A calls user B, the call is forwarded unconditional to user C, a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A.</td></tr>
<tr><td>Configuration</td><td>Subscription options:<br>• Connected user subscribed to COLR, Permanent = yes</td></tr>
<tr><td>SIP Parameter</td><td>200 OK<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>ANM<br>Redirection number restriction<br>Presentation restricted<br><br>--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow<br>SIP (Network A) | Interconnection Interface | SIP (Network B)<br>INVITE(Call-ID A-B) →<br>CFU is performed<br>← INVITE(Call-ID B-C), IAM<br>180 Ringing (Call-ID C-B) →<br>← 180 Ringing (Call-ID B-A, ACM)<br>200 OK INVITE (Call-ID C-B) →<br>← ACK (Call-ID B-C)<br>← 200 OK INVITE (Call-ID B-A; ANM)<br>ACK (Call-ID A-B) →<br>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.<br>Check: Is a 200 OK INVITE received at the interconnection interface?<br>Check: Is an ANM encapsulated in the 200 OK?<br>Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation restricted'?<br>Repeat this test in reverse direction.</td></tr>
</table>

| | |
|---|---|
| **Test case number** | SS_cfu_015 |
| **Test case group** | SIP-SIP/Service/CFU |
| **Reference** | 6.7/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 53 |
| **Test purpose** | **SIP-I support. CFU performed in Network B, No restriction of the Redirection number.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFU, Diverted-to user is not subscribed to the COLR service.<br>Ensure that when user A calls user B, the call is forwarded unconditional to user C, if a Redirection number restriction parameter is present it is set to 'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A. |
| **Configuration** | **Subscription options:**<br>• Connected user subscribed to COLR = no |
| **SIP Parameter** | 200 OK<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ANM**<br>Redirection number restriction<br>Presentation allowed<br>or<br>Redirection number restriction not present<br><br>--[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.<br>**Check:** Is a 200 OK INVITE received at the interconnection interface?<br>**Check:** Is an ANM encapsulated in the 200 OK?<br>**Check:** Is the ISUP/BICC Redirection number restriction present set to 'Presentation allowed' or is the parameter absent?<br>Repeat this test in reverse direction. |

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

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

#### 7.1.5.6.2 Communication Forwarding Busy (CFB)

| | |
|---|---|
| **Test case number** | SS_cfb_001 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 26 |
| **Test purpose** | **Communication forwarding busy, basic rules.**<br><br>The user A and user C are in Network A. The user B is in network B and is provided with CFB.<br>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.<br>**Check:** In the active call state, ensure the property of speech.<br>**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?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfb_002 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 26 AND SE 30 |
| **Test purpose** | **Communication forwarding busy, no notification.**<br><br>The user A and user C are in Network A. The user B is in network B and is provided with CFB, subscription option: "Originating user receives notification that his communication has been diverted" = No.<br>Ensure that when user A calls user B, the call is forwarded busy to user C, originating user is not notified. |
| **Configuration** | **Subscription options:**<br>• 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.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfb_003 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 26 AND SE 30 |
| **Test purpose** | **Communication forwarding busy, originating user is notified. URI from the served user not received.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFB "Originating user receives notification that his communication has been diverted" = Yes "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No. Ensure that when user A calls user B, the call is forwarded busy to user C, user A is notified of call diversion and not informed of the diverted-to number and served user number. |
| **Configuration** | **Subscription options:**<br>• Originating user receives notification that his communication has been diverted = Yes<br>• Served user allows the presentation of forwarded to URI to originating user in diversion notification = No<br>• Served user allows the presentation of his/her URI to originating user in diversion notification = No |
| **SIP Parameter** | 181 Being Forwarded<br>P-Asserted-Identity: <userB@NetworkB><br>Privacy: id<br>History-Info:<br><sip:userB@networkB?Privacy=history>;index=1,<br><sip: userC@networkA;cause=486?*Privacy=history*>;index=1.1 |
| **Message flow** | **SIP (Network A)** — **Interconnection Interface** — **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFB is performed**<br>← INVITE(Call-ID B-C)<br>← 181 Being Forwarded(Call-ID B-A)<br>180 Ringing(Call-ID C-B) →<br>← 180 Ringing(Call-ID B-A)<br>**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'.<br>**Check:** Is the cause parameter in the last entry set to '486'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**Check:** Is the P-Asserted-Identity header present in the 181 identifying the served user?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfb_004 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 26 AND SE 30 |
| **Test purpose** | **Communication forwarding busy, originating user is notified. URI from the diverted-to user received.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFB "Originating user receives notification that his communication has been diverted" = Yes "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes.<br>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:**<br>• Originating user receives notification that his communication has been diverted = Yes<br>• Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes |
| **SIP Parameter** | 181 Being Forwarded<br>P-Asserted-Identity: <userB@NetworkB><br>History-Info:<br>*<sip:userB@networkB>;index=1,*<br><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.<br>**Check:** A History-Info header is contained in the 181 with the URI of the diverted-to user.<br>**Check:** Is the cause parameter in the last entry set to '486'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**Check:** Is the P-Asserted-Identity header present in the 181 identifying the served user?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfb_005 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 26 AND SE 30 |
| **Test purpose** | **Communication forwarding busy, diverted-to user does not receive the URI of the served user.**<br><br>The user A and user C are in network C. The user B is in network B and is provided with CFB "Served user allows the presentation of his/her URI to the diverted-to user" = No.<br>Ensure that when user A calls user B, the call is forwarded busy "user determined user busy" to user C, user C is not informed of the forwarding number. |
| **Configuration** | **Subscription options:**<br>• Served user allows the presentation of his/her URI to the diverted-to user = No |
| **SIP Parameter** | INVITE:<br>History-Info:<br><sip:userB@networkB?Privacy=history&Reason=SIP%3Bcause%3D486><br>;index=1,<br><sip: userC@networkA;cause=486>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFB is performed**<br>← INVITE(Call-ID B-C)<br>**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'.<br>**Check:** Is the cause parameter in the last entry set to '486'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>**NOTE**: The "Reason" header in the first entry of the History-Info header sent to user C is only present if the call is forwarded due to "user determined user busy" by the served user.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfb_006 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 26 AND SE 30 |
| **Test purpose** | **Communication forwarding busy, diverted-to user receives the URI of the served user.**<br><br>The user A and user C are in network C. The user B is in network B and is provided with CFB "Served user allows the presentation of his/her URI to the diverted-to user" = Yes.<br>Ensure that when user A calls user B, the call is forwarded busy "user determined user busy" to user C, user C is informed of the forwarding number. |
| **Configuration** | **Subscription options:**<br>• Served user allows the presentation of his/her URI to the diverted-to user = Yes |
| **SIP Parameter** | INVITE:<br>History-Info:<br><sip:userB@networkB?Reason=SIP%3Bcause%3D486>;index=1,<br><sip: userC@networkA;cause=486>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFB is performed**<br>← INVITE(Call-ID B-C)<br>**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.<br>**Check:** Is the cause parameter in the last entry set to '486'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>**NOTE:** The "Reason" header in the first entry of the History-Info header sent to user C is only present if the call is forwarded due to "user determined user busy" by the served user.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfb_007 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 26 AND SE 30 |
| **Test purpose** | **Communication forwarding busy, full notification.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFB "Originating user receives notification that his communication has been diverted" = Yes "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes.<br>Ensure that when user A calls user B, the call is forwarded busy "user determined user busy" to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number. |
| **Configuration** | **Subscription options:**<br>• Originating user receives notification that his communication has been diverted = Yes<br>• Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes,<br>• diverting number is released to the diverted-to user = Yes |
| **SIP Parameter** | INVITE:<br>History-Info:<br><sip:userB@networkB&Reason=SIP%3Bcause%3D486>;index=1,<br><sip: userC@networkA;cause=486>;index=1.1<br><br>181 Being Forwarded<br>P-Asserted-Identity: <userB@NetworkB><br>History-Info:<br><sip:userB@networkB>;index=1,<br><sip: userC@networkA;cause=486>;index=1.1<br><br>200 OK INVITE<br>History-Info:<br><sip:userB@networkB&Reason=SIP%3B cause%3D486>;index=1,<br><sip: userC@networkA;cause=486>;index=1.1 |
| **Message flow** | **SIP (Network A)** — **Interconnection Interface** — **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFB is performed**<br>← INVITE(Call-ID B-C)<br>← 181 Being Forwarded(Call-ID B-A<br>180 Ringing(Call-ID C-B) →<br>← 180 Ringing(Call-ID B-A)<br>200 OK INVITE(Call-ID C-B) →<br>← ACK(Call-ID C-B)<br>← 200 OK INVITE(Call-ID B-A)<br>ACK(Call-ID A-B) →<br>**Communication**<br>**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.<br>**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.<br>**Check:** Is the cause parameter in the last entry set to '486'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>**NOTE:** The "Reason" header in the first entry of the History-Info header sent to user C is only present if the call is forwarded due to "user determined user busy" by the served user.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfb_008 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 26 |
| **Test purpose** | **Communication forwarding busy, unsuccessful UDUB.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFB.<br>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.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfb_009 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 26 |
| **Test purpose** | **Communication forwarding busy, unsuccessful NDUB.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFB.<br>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 A originating access.<br>**Check:** The dialogue is terminated by receiving a 486 Busy Here.<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_cfb_011 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 6.5/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55A |
| **Test purpose** | **SIP-I support. CFB performed in Network B, Notification subscription options is set to presentation not allowed.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number.<br>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.<br>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:**<br>• Calling user receives notification that his call has been diverted (forwarded or deflected) = no |
| **SIP Parameter** | 183 Session Progress<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ACM**<br>Backward call indicator<br>Called party's status indicator<br>no indication<br>Redirection number<br>Address signal (*Diverted-to user*)<br>Call diversion information<br>Notification subscription options<br>presentation not allowed<br>Redirecting reason<br>User Busy<br>Generic notification<br>call is diverting<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFB is performed**<br>← INVITE(Call-ID B-C, IAM)<br>← 183 Session Progress (Call-ID B-A, ACM)<br>**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.<br>**Check:** Is a 183 Session Progress received at the interconnection interface?<br>**Check:** Is an ACM encapsulated in the 183?<br>**Check:** Is the Called party's status indicator set to 'no indication'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Notification subscription options indicator set to 'presentation not allowed'?<br>**Check:** Is the Redirecting reason set to User Busy'?<br>Repeat this test in reverse direction. |

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

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

| | |
|---|---|
| **Test case number** | SS_cfb_014 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 6.7/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55A |
| **Test purpose** | **SIP-I support. CFB performed in Network B, Restriction of the Redirection number.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Diverted-to user is subscribed to the COLR service in Permanent mode.<br>Ensure that when user A calls user B, the call is forwarded on busy user to user C, a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A. |
| **Configuration** | **Subscription options:**<br>• Connected user subscribed to COLR, Permanent = yes |
| **SIP Parameter** | 200 OK<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ANM**<br>Redirection number restriction<br>Presentation restricted<br><br>--[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<br>**Check:** Is a 200 OK INVITE received at the interconnection interface?<br>**Check:** Is an ANM encapsulated in the 200 OK?<br>**Check:** Is the ISUP/BICC Redirection number restriction set to 'Presentation restricted'?<br>Repeat this test in reverse direction. |

<table>
<tr><td>Test case number</td><td>SS_cfb_015</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/CFB</td></tr>
<tr><td>Reference</td><td>6.7/ [24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>[Network B] SE 17 AND [Network B] SE 47 AND SE 55A</td></tr>
<tr><td>Test purpose</td><td>SIP-I support. CFB performed in Network B, No restriction of the Redirection number.<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Diverted-to user is not subscribed to the COLR service.<br>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.</td></tr>
<tr><td>Configuration</td><td>Subscription options:<br>• Connected user subscribed to COLR = no</td></tr>
<tr><td>SIP Parameter</td><td>200 OK<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>ANM<br>Redirection number restriction<br>Presentation allowed<br>or<br>Redirection number restriction not present<br><br>--[any boundary name]--</td></tr>
<tr><td colspan="2">Message flow<br><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.<br>Check: Is a 200 OK INVITE received at the interconnection interface?<br>Check: Is an ANM encapsulated in the 200 OK?<br>Check: Is the ISUP/BICC Redirection number restriction present set to 'Presentation allowed' or is the parameter absent?<br>Repeat this test in reverse direction.</td></tr>
</table>

| | |
|---|---|
| **Test case number** | SS_cfb_016 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55A |
| **Test purpose** | **SIP-I support. CFB performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Served user releases his/her number to diverted-to user = Release diverting number information.<br>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.<br>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:**<br>• Served user releases his/her number to diverted-to user = Release diverting number information |
| **SIP Parameter** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>Redirecting number<br>Address presentation restricted indicator<br>presentation allowed<br>Address signal (*Diverting user*)<br>Original called number<br>Address presentation restricted indicator<br>presentation allowed<br>Address signal<br>Redirection information<br>Original Redirection Reason<br>unknown<br>Redirecting indicator<br>Redirection counter<br>Redirecting reason<br>User Busy<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFB is performed**<br>← INVITE(Call-ID B-C, IAM)<br>**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.<br>**Check:** **Is an INVITE request received at the interconnection interface**?<br>**Check:** Is an IAM encapsulated in the INVITE?<br>**Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'?<br>**Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Redirection information present and the Redirecting reason is set to 'User Busy'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfb_017 |
| **Test case group** | SIP-SIP/Service/CFB |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55A |
| **Test purpose** | **SIP-I support. CFB performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFB, Served user releases his/her number to diverted-to user = Release diverting number information.<br>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.<br>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:<br>• Served user releases his/her number to diverted-to user = Do not release diverting numberinformation |
| **SIP Parameter** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>Redirecting number<br>Address presentation restricted indicator<br>presentation restricted<br>Address signal (*Diverting user*)<br>Original called number<br>Address presentation restricted indicator<br>presentation restricted<br>Address signal<br>Redirection information<br>Original Redirection Reason<br>unknown<br>Redirecting indicator<br>Redirection counter<br>Redirecting reason<br>User Busy<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFB is performed**<br>← INVITE(Call-ID B-C, IAM)<br>**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.<br>**Check:** **Is an INVITE request received at the interconnection interface**?<br>**Check:** Is an IAM encapsulated in the INVITE?<br>**Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation restricted'?<br>**Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation restricted'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Redirection information present and the Redirecting reason is set to 'User Busy'?<br>Repeat this test in reverse direction. |

### 7.1.5.6.3 Communication Forwarding No Reply (CFNR)

| | |
|---|---|
| **Test case number** | SS_cfnr_001 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 27 |
| **Test purpose** | **Communication forwarding no reply, basic rules.**<br><br>The user A and user C are in Network A. The user B is in network B and is provided with CFNR.<br>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.<br>**Check:** In the active call state, ensure the property of speech.<br>**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?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnr_002 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 27 AND SE 30 |
| **Test purpose** | **Communication forwarding no reply, no notification.**<br><br>The user A and user C are in Network A. The user B is in network B and is provided with CFNR, subscription option: "Originating user receives notification that his communication has been diverted" = No.<br>Ensure that when user A calls user B, the call is forwarded no reply to user C, originating user is not notified. |
| **Configuration** | **Subscription options:**<br>• 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.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnr_003 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 27 AND SE 30 |
| **Test purpose** | **Communication forwarding no reply, originating user is notified. URI from the served user not received.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFNR "Originating user receives notification that his communication has been diverted" = Yes "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No.<br>Ensure that when user A calls user B, the call is forwarded no reply to user C, user A is notified of call diversion and not informed of the diverted-to number and served user number. |
| **Configuration** | **Subscription options:**<br>• Originating user receives notification that his communication has been diverted = Yes<br>• Served user allows the presentation of forwarded to URI to originating user in diversion notification = No<br>• Served user allows the presentation of his/her URI to originating user in diversion notification = No |
| **SIP Parameter** | 181 Being Forwarded<br>P-Asserted-Identity: <userB@NetworkB><br>Privacy: id<br>History-Info:<br><sip:userB@networkB?Privacy=history>;index=1,<br><sip: userC@networkA;cause=408?Privacy=history>;index=1.1 |
| **Message flow** | **SIP (Network A)** \| **Interconnection Interface** \| **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing(Call-ID B-A)<br>**CFB is performed**<br>← INVITE(Call-ID B-C)<br>← 181 Being Forwarded(Call-ID B-A)<br>180 Ringing(Call-ID C-B) →<br>← 180 Ringing(Call-ID B-A)<br>**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'.<br>**Check:** Is the cause parameter in the last entry set to '408'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**Check:** Is the P-Asserted-Identity header present in the 181 identifying the served user?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnr_004 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 27 AND SE 30 |
| **Test purpose** | **Communication forwarding no reply, originating user is notified. URI from the diverted-to user received.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFNR "Originating user receives notification that his communication has been diverted" = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes.<br>Ensure that when user A calls user B, the call is forwarded no reply to user C, user A is notified of call diversion and informed of the diverted-to number. |
| **Configuration** | **Subscription options:**<br>• Originating user receives notification that his communication has been diverted = Yes<br>• Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes |
| **SIP Parameter** | 181 Being Forwarded<br>P-Asserted-Identity: <userB@NetworkB><br>History-Info:<br><sip:userB@networkB>;index=1,<br><sip: userC@networkA;cause=408>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** — **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing(Call-ID B-A)<br>**CFB is performed**<br>← INVITE(Call-ID B-C)<br>← 181 Being Forwarded(Call-ID B-A)<br>180 Ringing(Call-ID C-B) →<br>← 180 Ringing(Call-ID B-A)<br>**Apply post test routine** |
| **Comments** | **Check:** A 181 Being Forwarded is received at the interconnection interface.<br>**Check:** A History-Info header is contained in the 181 with the URI of the diverted-to user.<br>**Check:** Is the cause parameter in the last entry is set to '408'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**Check:** Is the P-Asserted-Identity header present in the 181 identifying the served user?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnr_005 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 27 AND SE 30 |
| **Test purpose** | **Communication forwarding no reply, diverted-to user does not receive the URI of the served user.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with "Served user allows the presentation of his/her URI to the diverted-to user" = No.<br>Ensure that when user A calls user B, the call is forwarded no reply to user C, user C is not informed of the forwarding number. |
| **Configuration** | **Subscription options:**<br>• Served user allows the presentation of his/her URI to the diverted-to user = No |
| **SIP Parameter** | INVITE<br>History-Info:<br><sip:userB@networkB?Privacy=history>;index=1,<br><sip: userC@network1;cause=408>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing(Call-ID B-A)<br>**CFB is performed**<br>← INVITE(Call-ID B-C)<br>**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'.<br>**Check:** Is the cause parameter in the last entry is set to '408'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnr_006 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 27 AND SE 30 |
| **Test purpose** | **Communication forwarding no reply, diverted-to user receives the URI of the diverted-to user.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with "Served user allows the presentation of his/her URI to the diverted-to user" = Yes.<br>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:**<br>• Served user allows the presentation of his/her URI to the diverted-to user = Yes |
| **SIP Parameter** | INVITE<br>History-Info:<br><sip:userB@networkB>;index=1,<br><sip: userC@network1;cause=408>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** / **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing(Call-ID B-A)<br>**CFB is performed**<br>← INVITE(Call-ID B-C)<br>**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.<br>**Check:** Is the 'cause' parameter present in the Request line sent to user C (diverted-to user) set to '408'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**Check:** Is the P-Asserted-Identity header present in the 181 identifying the served user?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnr_007 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 27 AND SE 30 |
| **Test purpose** | **Communication forwarding no reply, full notification.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFNR "Originating user receives notification that his communication has been diverted" = Yes, "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes.<br>Ensure that when user A calls user B, the call is forwarded no reply to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number. |
| **Configuration** | **Subscription options:**<br>• Originating user receives notification that his communication has been diverted = Yes<br>• Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes<br>• diverting number is released to the diverted-to user = Yes |
| **SIP Parameter** | INVITE:<br>History-Info:<br><sip:userB@networkB>;index=1,<br><sip: userC@networkA;cause=486>;index=1.1<br><br>181 Being Forwarded<br>P-Asserted-Identity: <userB@NetworkB><br>History-Info<br><sip:userB@network>;index=1,<br><sip: userC@networkA;cause=408>;index=1.1<br><br>200 OK INVITE<br>History-Info:<br><sip:userB@networkB>;index=1,<br><sip: userC@networkA;cause=408>;index=1.1 |
| **Message flow** | **SIP (Network A)** — **Interconnection Interface** — **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing(Call-ID B-A)<br>**CFB is performed**<br>← INVITE(Call-ID B-C)<br>← 181 Being Forwarded(Call-ID B-A<br>180 Ringing(Call-ID C-B) →<br>← 180 Ringing(Call-ID B-A)<br>200 OK INVITE(Call-ID C-B) →<br>← ACK(Call-ID C-B)<br>← 200 OK INVITE(Call-ID B-A)<br>ACK(Call-ID A-B) →<br>**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.<br>**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.<br>**Check:** Is the cause parameter in the last entry is set to '408'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**Check:** Is the P-Asserted-Identity header present in the 181 identifying the served user?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnr_008 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 27 |
| **Test purpose** | **Communication forwarding no reply, unsuccessful UDUB.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFNR.<br>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)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing(Call-ID B-A)<br>**CFB is performed**<br>← INVITE(Call-ID B-C)<br>486 Busy Here(Call-ID C-B) →<br>← ACK(Call-ID B-C)<br>← 486 Busy Here(Call-ID A-B)<br>ACK(Call-ID A-B) → |
| **Comments** | **Check:** The dialogue is terminated by receiving a 486 Busy Here.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnr_009 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 27 |
| **Test purpose** | **Communication forwarding no reply, unsuccessful NDUB.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFNR.<br>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)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing(Call-ID B-A)<br>**CFB is performed**<br>← INVITE(Call-ID B-C)<br>486 Busy Here(Call-ID C-B) →<br>← ACK(Call-ID B-C)<br>← 486 Busy Here(Call-ID A-B)<br>ACK(Call-ID A-B) → |
| **Comments** | **Check:** The dialogue is terminated by receiving a 486 Busy Here.<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_cfnr_011 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 6.5/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55B |
| **Test purpose** | **SIP-I support. CFNR performed in Network B, Notification subscription options is set to presentation not allowed.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number.<br>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.<br>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:**<br>• Calling user receives notification that his call has been diverted (forwarded or deflected) = no |
| **SIP Parameter** | 183 Session Progress<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**CPG**<br>Event indicator<br>Alerting or Progress<br>Redirection number<br>Address signal (*Diverted-to user*)<br>Call diversion information<br>Notification subscription options<br>presentation not allowed<br>Redirecting reason<br>No reply<br>Generic notification<br>call is diverting<br><br>--[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.<br>**Check:** Is a 183 Session Progress received at the interconnection interface?<br>**Check:** Is a CPG encapsulated in the 183?<br>**Check:** Is the Called party's status indicator set to 'no indication'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Notification subscription options indicator set to 'presentation not allowed'?<br>**Check:** Is the Redirecting reason set to 'No reply'?<br>Repeat this test in reverse direction. |

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

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

<table>
<tr><td>Test case number</td><td colspan="3">SS_cfnr_014</td></tr>
<tr><td>Test case group</td><td colspan="3">SIP-SIP/Service/CFNR</td></tr>
<tr><td>Reference</td><td colspan="3">6.7/ [24]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td colspan="3">[Network B] SE 17 AND [Network B] SE 47 AND SE 53B</td></tr>
<tr><td>Test purpose</td><td colspan="3">SIP-I support. CFNR performed in Network B, Restriction of the Redirection number.<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Diverted-to user is subscribed to the COLR service in Permanent mode.<br>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.</td></tr>
<tr><td>Configuration</td><td colspan="3">Subscription options:<br>• Connected user subscribed to COLR, Permanent = yes</td></tr>
<tr><td>SIP Parameter</td><td colspan="3">200 OK<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>ANM<br>Redirection number restriction<br>Presentation restricted<br><br>--[any boundary name]--</td></tr>
<tr><td colspan="4">Message flow</td></tr>
<tr><td>SIP (Network A)</td><td></td><td>Interconnection Interface</td><td>SIP (Network B)</td></tr>
<tr><td></td><td></td><td>INVITE(Call-ID A-B)</td><td>→</td></tr>
<tr><td></td><td>←</td><td>180 Ringing (Call-ID B-A, ACM)</td><td></td></tr>
<tr><td></td><td></td><td>CFNR is performed</td><td></td></tr>
<tr><td></td><td>←</td><td>INVITE(Call-ID B-C, IAM)</td><td></td></tr>
<tr><td></td><td></td><td>180 Ringing (Call-ID C-B)</td><td>→</td></tr>
<tr><td></td><td>←</td><td>180 Ringing (Call-ID B-A, ACM)</td><td></td></tr>
<tr><td></td><td></td><td>200 OK INVITE (Call-ID C-B)</td><td>→</td></tr>
<tr><td></td><td>←</td><td>ACK (Call-ID B-C)</td><td></td></tr>
<tr><td></td><td>←</td><td>200 OK INVITE (Call-ID B-A; ANM)</td><td></td></tr>
<tr><td></td><td></td><td>ACK (Call-ID A-B)</td><td>→</td></tr>
<tr><td></td><td></td><td>Apply post test routine</td><td></td></tr>
<tr><td>Comments</td><td colspan="3">Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A.<br>Check: Is a 200 OK INVITE received at the interconnection interface?<br>Check: Is an ANM encapsulated in the 200 OK?<br>Check: Is the ISUP/BICC Redirection number restriction set to 'Presentation restricted'?<br>Repeat this test in reverse direction.</td></tr>
</table>

| | |
|---|---|
| **Test case number** | SS_cfnr_015 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 6.7/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55B |
| **Test purpose** | **SIP-I support. CFNR performed in Network B, No restriction of the Redirection number.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Diverted-to user is not subscribed to the COLR service.<br>Ensure that when user A calls user B, the call is forwarded on no reply to user C, if a Redirection number restriction parameter is present it is set to 'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A. |
| **Configuration** | **Subscription options:**<br>• Connected user subscribed to COLR = no |
| **SIP Parameter** | 200 OK<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ANM**<br>Redirection number restriction<br>Presentation allowed<br>or<br>Redirection number restriction not present<br><br>--[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.<br>**Check:** Is a 200 OK INVITE received at the interconnection interface?<br>**Check:** Is an ANM encapsulated in the 200 OK?<br>**Check:** Is the ISUP/BICC Redirection number restriction present set to 'Presentation allowed' or is the parameter absent?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnr_016 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55B |
| **Test purpose** | **SIP-I support. CFNR performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Served user releases his/her number to diverted-to user = Release diverting number information.<br>Ensure that when user A calls user B, the call is forwarded on no reply to user C, user C is notified of call diversion and informed of the diverting number.<br>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:**<br>• Served user releases his/her number to diverted-to user = Release diverting number information |
| **SIP Parameter** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>Redirecting number<br>Address presentation restricted indicator<br>presentation allowed<br>Address signal (*Diverting user*)<br>Original called number<br>Address presentation restricted indicator<br>presentation allowed<br>Address signal<br>Redirection information<br>Original Redirection Reason<br>unknown<br>Redirecting indicator<br>Redirection counter<br>Redirecting reason<br>No reply<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing (Call-ID B-A, ACM)<br>**CFNR is performed**<br>← INVITE(Call-ID B-C, IAM)<br>**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.<br>**Check:** **Is an INVITE request received at the interconnection interface**?<br>**Check:** Is an IAM encapsulated in the INVITE?<br>**Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'?<br>**Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Redirection information present and the Redirecting reason is set to 'No reply'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnr_017 |
| **Test case group** | SIP-SIP/Service/CFNR |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55B |
| **Test purpose** | **SIP-I support. CFNR performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CFNR, Served user releases his/her number to diverted-to user = Release diverting number information.<br>Ensure that when user A calls user B, the call is forwarded on no reply to user C, user C is notified of call diversion and informed of the diverting number.<br>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:<br>• Served user releases his/her number to diverted-to user = Do not release diverting numberinformation |
| **SIP Parameter** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>Redirecting number<br>Address presentation restricted indicator<br>presentation restricted<br>Address signal (*Diverting user*)<br>Original called number<br>Address presentation restricted indicator<br>presentation restricted<br>Address signal<br>Redirection information<br>Original Redirection Reason<br>unknown<br>Redirecting indicator<br>Redirection counter<br>Redirecting reason<br>No reply<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing (Call-ID B-A, ACM)<br>**CFNR is performed**<br>← INVITE(Call-ID B-C, IAM)<br>**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.<br>**Check:** **Is an INVITE request received at the interconnection interface**?<br>**Check:** Is an IAM encapsulated in the INVITE?<br>**Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation restricted'?<br>**Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation restricted'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Redirection information present and the Redirecting reason is set to 'No reply'?<br>Repeat this test in reverse direction. |

#### 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/ [9] |
| **SELECTION EXPRESSION** | SE 28 |
| **Test purpose** | **Communication forwarding not logged in, basic rules.**<br><br>The user A and user C are in Network A. The user B is in network B and is provided with CFNL.<br>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**<br>**SIP (Network A)** | **Interconnection Interface** — **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFNL is performed**<br>← INVITE(Call-ID B-C)<br>180 Ringing(Call-ID C-B) →<br>← 180 Ringing(Call-ID B-A)<br>200 OK INVITE(Call-ID C-B) →<br>← ACK(Call-ID B-C)<br>← 200 OK INVITE(Call-ID B-A)<br>ACK(Call-ID A-B) →<br>**Communication**<br>**Apply post test routine** |
| **Comments** | **Check:** The CDIV not logged in is successful.<br>**Check:** In the active call state, ensure the property of speech.<br>**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?<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_cfnl_003 |
| **Test case group** | SIP-SIP/Service/CFNL |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 28 AND SE 30 |
| **Test purpose** | **Communication forwarding not logged in, originating user is notified. URI of the diverted-to user not received.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFNL "Originating user receives notification that his communication has been diverted" = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No.<br>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:**<br>• Originating user receives notification that his communication has been diverted = Yes<br>• Served user allows the presentation of forwarded to URI to originating user in diversion notification = No<br>• Served user allows the presentation of his/her URI to originating user in diversion notification = No |
| **SIP Parameter** | 181 Being Forwarded<br>P-Asserted-Identity: <userB@NetworkB><br>Privacy: id<br>History-Info:<br>*<sip:userB@networkB?Privacy=history>;index=1*,<br>*<sip: userC@networkA;cause=404?Privacy=history>;index=1.1* |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFNL is performed**<br>← INVITE(Call-ID B-C)<br>← 181 Being Forwarded(Call-ID B-A)<br>180 Ringing(Call-ID C-B) →<br>← 180 Ringing(Call-ID B-A)<br>**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'.<br>**Check:** Is the cause parameter in the last entry is set to '404'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**Check:** Is the P-Asserted-Identity header present in the 181 identifying the served user?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>Repeat this test in reverse direction. |

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

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|---|---|
| **Test case number** | SS_cfnl_005 |
| **Test case group** | SIP-SIP/Service/CFNL |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 28 AND SE 30 |
| **Test purpose** | **Communication forwarding not logged in, diverted-to user does not receive the URI of the diverted-to user.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFNL "Served user allows the presentation of his/her URI to diverted-to user" = No.<br>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:**<br>• Served user allows the presentation of his/her URI to diverted-to user = No |
| **SIP Parameter** | INVITE<br>History-Info:<br><sip:userB@networkB?Privacy=history>;index=1,<br><sip: userC@network1;cause=404>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFNL is performed**<br>← INVITE(Call-ID B-C)<br>**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'.<br>**Check:** Is the cause parameter in the last entry is set to '404'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnl_006 |
| **Test case group** | SIP-SIP/Service/CFNL |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 28 AND SE 30 |
| **Test purpose** | **Communication forwarding not logged in, diverted-to user receives the URI of the served user.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFNL "Served user allows the presentation of his/her URI to diverted-to user" = Yes.<br>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:**<br>• Served user allows the presentation of his/her URI to diverted-to user = Yes |
| **SIP Parameter** | INVITE<br>History-Info:<br>*<sip:userB@networkB>;index=1*,<br><sip: userC@networkA;cause=404>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFNL is performed**<br>← INVITE(Call-ID B-C)<br>**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.<br>**Check:** Is the cause parameter in the last entry is set to '404'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnl_007 |
| **Test case group** | SIP-SIP/Service/CFNL |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 28 AND SE 30 |
| **Test purpose** | **Communication forwarding not logged in, full notification.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFNL "Originating user receives notification that his communication has been diverted" = Yes, "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes, "diverting number is released to the diverted-to user" = Yes.<br>Ensure that when user A calls user B, the call is forwarded not logged in to user C, user A is notified of call diversion and informed of the diverted-to number and user C is informed of the forwarding number. |
| **Configuration** | **Subscription options:**<br>• Originating user receives notification that his communication has been diverted = Yes<br>• Served user allows the presentation of forwarded to URI to originating user in diversion notification = Yes<br>• diverting number is released to the diverted-to user = Yes |
| **SIP Parameter** | INVITE:<br>History-Info:<br><sip:userB@networkB&Reason=SIP;cause=404>;index=1,<br><sip: userC@networkA;cause=404>;index=1.1<br><br>181 Being Forwarded<br>P-Asserted-Identity: <userB@NetworkB><br>History-Info:<br><sip:userB@network>;index=1,<br><sip: userC@networkA;cause=404>;index=1.1<br><br>200 OK INVITE<br>History-Info:<br><sip:userB@networkB>;index=1,<br><sip: userC@networkA;cause=404>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFNL is performed**<br>← INVITE(Call-ID B-C)<br>← 181 Being Forwarded(Call-ID B-A<br>180 Ringing(Call-ID C-B) →<br>← 180 Ringing(Call-ID B-A)<br>200 OK INVITE(Call-ID C-B) →<br>← ACK(Call-ID C-B)<br>← 200 OK INVITE(Call-ID B-A)<br>ACK(Call-ID A-B) →<br>**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.<br>**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.<br>**Check:** Is the cause parameter in the last entry is set to '404'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**Check:** Is the P-Asserted-Identity header present in the 181 identifying the served user?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

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|---|---|
| **Test case number** | SS_cfnl_008 |
| **Test case group** | SIP-SIP/Service/CFNL |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 28 |
| **Test purpose** | **Communication forwarding not logged in, unsuccessful UDUB.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFNL.<br>Ensure that when user A calls user B, the call is forwarded not logged in to user C and user C is user determined user busy. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow** | |

| **SIP (Network A)** | | **Interconnection Interface** | | **SIP (Network B)** |
|---|---|---|---|---|
| | | INVITE(Call-ID A-B) | → | |
| | | **CFNL is performed** | | |
| | | 486 Busy Here(Call-ID C-B) | → | |
| | ← | ACK(Call-ID B-C) | | |
| | ← | 486 Busy Here(Call-ID A-B) | | |
| | | ACK(Call-ID A-B) | → | |

| | |
|---|---|
| **Comments** | **Check:** The dialogue is terminated by receiving a 486 Busy Here.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnl_009 |
| **Test case group** | 4.5.2.6/ [9] |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 28 |
| **Test purpose** | **Communication forwarding not logged in, unsuccessful NDUB.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CFNL.<br>Ensure that when user A calls user B, the call is forwarded not logged in to user C and user C is busy. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow** | |

| **SIP (Network A)** | | **Interconnection Interface** | | **SIP (Network B)** |
|---|---|---|---|---|
| | | INVITE(Call-ID A-B) | → | |
| | | **CFNL is performed** | | |
| | | 486 Busy Here(Call-ID C-B) | → | |
| | ← | ACK(Call-ID B-C) | | |
| | ← | 486 Busy Here(Call-ID A-B) | | |
| | | ACK(Call-ID A-B) | → | |

| | |
|---|---|
| **Comments** | **Check:** The dialogue is terminated by receiving a 486 Busy Here.<br>Repeat this test in reverse direction. |

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

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

| | |
|---|---|
| **Test case number** | SS_cfnl_012 |
| **Test case group** | SIP-SIP/Service/CFNL |
| **Reference** | 6.5/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55C |
| **Test purpose** | **SIP-I support. Mobile subscriber not reachable performed in Network B, Notification subscription options is set to presentation allowed without redirection number.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with Mobile subscriber not reachable, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number.<br>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.<br>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:**<br>• Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number |
| **SIP Parameter** | 183 Session Progress<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ACM**<br>Backward call indicator<br>Called party's status indicator<br>no indication<br>Redirection number<br>Address signal (*Diverted-to user*)<br>Call diversion information<br>Notification subscription options<br>presentation allowed without redirection number<br>Redirecting reason<br>Mobile subscriber not reachable<br>Generic notification<br>call is diverting<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFNL is performed**<br>← INVITE(Call-ID B-C, IAM)<br>← 183 Session Progress (Call-ID B-A, ACM)<br>**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.<br>**Check:** 183 Session Progress is received at the interconnection interface.<br>**Check:** Is an ACM encapsulated in the 183?<br>**Check:** Is the Called party's status indicator set to 'no indication'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Notification subscription options indicator is set to 'presentation allowed without redirection number'?<br>**Check:** Is the Redirecting reason set to 'Mobile subscriber not reachable'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnl_013 |
| **Test case group** | SIP-SIP/Service/CFNL |
| **Reference** | 6.5/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55C |
| **Test purpose** | **SIP-I support. Mobile subscriber not reachable performed in Network B, Notification subscription options is set to presentation allowed with redirection number.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with Mobile subscriber not reachable, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number.<br>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.<br>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:**<br>• Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number |
| **SIP Parameter** | 183 Session Progress<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ACM**<br>Backward call indicator<br>Called party's status indicator<br>no indication<br>Redirection number<br>Address signal (*Diverted-to user*)<br>Call diversion information<br>Notification subscription options<br>presentation allowed with redirection number<br>Redirecting reason<br>Mobile subscriber not reachable<br>Generic notification<br>call is diverting<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFNL is performed**<br>← INVITE(Call-ID B-C, IAM)<br>← 183 Session Progress (Call-ID B-A, ACM)<br>**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.<br>**Check:** 183 Session Progress is received at the interconnection interface<br>**Check:** Is an ACM encapsulated in the 183?<br>**Check:** Is the Called party's status indicator set to 'no indication'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Notification subscription options indicator is set to 'presentation allowed with redirection number'?<br>**Check:** Is the Redirecting reason set to 'Mobile subscriber not reachable'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnl_014 |
| **Test case group** | SIP-SIP/Service/CFNL |
| **Reference** | 6.7/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 53 AND SE 55C |
| **Test purpose** | **SIP-I support. Mobile subscriber not reachable performed in Network B, Restriction of the Redirection number.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with Mobile subscriber not reachable, Diverted-to user is subscribed to the COLR service in Permanent mode.<br>Ensure that when user A calls user B, the call is forwarded not logged in to user C, a Redirection number restriction parameter is present set to 'Presentation restricted' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A. |
| **Configuration** | **Subscription options:**<br>• Connected user subscribed to COLR, Permanent = yes |
| **SIP Parameter** | 200 OK<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ANM**<br>Redirection number restriction<br>Presentation restricted<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFNL is performed**<br>← INVITE(Call-ID B-C, IAM)<br>180 Ringing (Call-ID C-B) →<br>← 180 Ringing (Call-ID B-A, ACM)<br>200 OK INVITE (Call-ID C-B) →<br>← ACK (Call-ID B-C)<br>← 200 OK INVITE (Call-ID B-A, ANM)<br>ACK (Call-ID A-B) →<br>**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.<br>**Check:** Is a 200 OK INVITE received at the interconnection interface<br>**Check:** Is an ANM encapsulated in the 200 OK?<br>**Check:** Is the ISUP/BICC Redirection number restriction set to 'Presentation restricted'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnl_015 |
| **Test case group** | SIP-SIP/Service/CFNL |
| **Reference** | 6.7/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 53 AND SE 55C |
| **Test purpose** | **SIP-I support. Mobile subscriber not reachable performed in Network B, No restriction of the Redirection number.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with Mobile subscriber not reachable, Diverted-to user is not subscribed to the COLR service.<br>Ensure that when user A calls user B, the call is forwarded not logged in to user C, if a Redirection number restriction parameter is present it is set to 'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A. |
| **Configuration** | **Subscription options:**<br>• Connected user subscribed to COLR = no |
| **SIP Parameter** | 200 OK<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ANM**<br>Redirection number restriction<br>Presentation allowed<br>or<br>Redirection number restriction not present<br><br>--[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.<br>**Check:** Is a 200 OK INVITE received at the interconnection interface?<br>**Check:** Is an ANM encapsulated in the 200 OK?<br>**Check:** Is the ISUP/BICC Redirection number restriction present set to 'Presentation allowed' or is the parameter absent?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnl_016 |
| **Test case group** | SIP-SIP/Service/CFNL |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55C |
| **Test purpose** | **SIP-I support. Mobile subscriber not reachable performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with Mobile subscriber not reachable, Served user releases his/her number to diverted-to user = Release diverting number information.<br>Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user C is notified of call diversion and informed of the diverting number.<br>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:**<br>• Served user releases his/her number to diverted-to user = Release diverting number information |
| **SIP Parameter** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>Redirecting number<br>Address presentation restricted indicator<br>presentation allowed<br>Address signal (*Diverting user*)<br>Original called number<br>Address presentation restricted indicator<br>presentation allowed<br>Address signal<br>Redirection information<br>Original Redirection Reason<br>unknown<br>Redirecting indicator<br>Redirection counter<br>Redirecting reason<br>Mobile subscriber not reachable<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** / **SIP (Network B)**<br>INVITE(Call-ID A-B) ➔<br>**CFNL is performed**<br>🡐 INVITE(Call-ID B-C, IAM)<br>**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.<br>**Check:** Is an INVITE request received at the interconnection interface?<br>**Check:** Is an IAM encapsulated in the INVITE?<br>**Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'?<br>**Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Redirection information present and the Redirecting reason is set to 'Mobile subscriber not reachable'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cfnl_017 |
| **Test case group** | SIP-SIP/Service/CFNL |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55C |
| **Test purpose** | **SIP-I support. Mobile subscriber not reachable performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with Mobile subscriber not reachable, Served user releases his/her number to diverted-to user = Release diverting number information.<br>Ensure that when user A calls user B, the call is forwarded on Mobile subscriber not reachable to user C, user C is notified of call diversion and informed of the diverting number.<br>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:<br>• Served user releases his/her number to diverted-to user = Do not release diverting numberinformation |
| **SIP Parameter** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>Redirecting number<br>Address presentation restricted indicator<br>presentation restricted<br>Address signal (*Diverting user*)<br>Original called number<br>Address presentation restricted indicator<br>presentation restricted<br>Address signal<br>Redirection information<br>Original Redirection Reason<br>unknown<br>Redirecting indicator<br>Redirection counter<br>Redirecting reason<br>Mobile subscriber not reachable<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CFNL is performed**<br>← INVITE(Call-ID B-C, IAM)<br>**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.<br>**Check:** Is an INVITE request received at the interconnection interface?<br>**Check:** Is an IAM encapsulated in the INVITE?<br>**Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation restricted'?<br>**Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation restricted'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Redirection information present and the Redirecting reason is set to 'Mobile subscriber not reachable'?<br>Repeat this test in reverse direction. |

### 7.1.5.6.5 Communication Deflection

| | |
|---|---|
| **Test case number** | SS_cd_001 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 29 |
| **Test purpose** | **Communication deflection during alerting, basic rules.**<br><br>The user A and user C are in Network A. The user B is in network B and is provided with CDa.<br>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.<br>**Check:** In the active call state, ensure the property of speech.<br>**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?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cd_002 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 29 |
| **Test purpose** | **Communication deflection immediate, basic rules.**<br><br>The user A and user C are located in Network A. The user B is located in network B and is provided with CDi. Ensure that when user A calls user B which deflects immediately the communication towards user C (i.e. before alerting starts), the call is forwarded to user C. In the active call state, ensure the property of speech. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow** | |

| SIP (Network A) | | Interconnection Interface | | 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.<br>**Check:** In the active call state, ensure the property of speech.<br>**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?<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_cd_004 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 29 AND SE 30 |
| **Test purpose** | **Communication Deflection immediate response, originating user is notified. URI of the diverted-to user not received.**<br><br>The user A and user C are located in network A. The user B is located in network B and is provided with CDi "Originating user receives notification that his communication has been diverted" = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = No and. "Served user allows the presentation of his/her URI to originating user in diversion notification" = No.<br>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.<br>Ensure that User A receives a 181 Call Is Being Forwarded message, user A is notified of call diversion and not informed of the diverted-to number and served user number. |
| **Configuration** | **Subscription options:**<br>• Originating user receives notification that his communication has been diverted = Yes<br>• Originating user receives notification that his communication has been diverted = No<br>• Served user allows the presentation of his/her URI to originating user in diversion notification = No |
| **SIP Parameter** | 181 Being Forwarded<br>P-Asserted-Identity: <userB@NetworkB><br>Privacy: id<br>History-Info:<br><sip:userB@networkB?Privacy=history>;index=1,<br><sip: userC@networkA;cause=480?Privacy=history>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CDi is performed**<br>← INVITE(Call-ID B-C)<br>← 181 Being Forwarded(Call-ID B-A)<br>**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'.<br>**Check:** Is the cause parameter in the last entry is set to '480'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**Check:** Is the P-Asserted-Identity header present in the 181 identifying the served user?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cd_005 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 29 AND SE 30 |
| **Test purpose** | **Communication Deflection immediate response, originating user is notified. URI from the diverted-to user received.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CDi "Originating user receives notification that his communication has been diverted" = Yes and "Served user allows the presentation of forwarded to URI to originating user in diversion notification" = Yes.<br>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.<br>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:**<br>• Originating user receives notification that his communication has been diverted = Yes<br>• Served user allows the presentation of diverted to URI to originating user in diversion notification = Yes |
| **SIP Parameter** | 181 Being Forwarded<br>P-Asserted-Identity: <userB@NetworkB><br>History-Info:<br><sip:userB@networkB>;index=1,<br><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.<br>**Check:** A History-Info header is contained in the 181 with the URI of the diverted-to user.<br>**Check:** Is the cause parameter in the last entry is set to '480'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**Check:** Is the P-Asserted-Identity header present in the 181 identifying the served user?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cd_006 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 29 AND SE 30 |
| **Test purpose** | **Communication Deflection immediate response, diverted-to user does not receive the URI of the served user.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CDi "Served user allows the presentation of his/her URI to the diverted-to user" = No.<br>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:**<br>• Served user allows the presentation of his/her URI to diverted-to user = No |
| **SIP Parameter** | INVITE<br>History-Info:<br><sip:userB@networkB?Privacy=history&Reason=SIP%3Bcause%3D302><br>;index=1,<br><sip: userC@networkA;cause=480>;index=1.1 |
| **Message flow** | |
| **SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CDi is performed**<br>← INVITE(Call-ID B-C)<br>**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'.<br>**Check:** Is the cause parameter in the last entry is set to '480'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cd_007 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 29 AND SE 30 |
| **Test purpose** | **Communication Deflection immediate response, diverted-to user receives the URI of the served user.**<br><br>The user A and user C are in network A. The user B is in network B and is provided with CDi "Served user allows the presentation of his/her URI to diverted-to user" = Yes.<br>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:**<br>• Served user allows the presentation of his/her URI to diverted-to user = Yes |
| **SIP Parameter** | INVITE<br>History-Info:<br><sip:userB@networkB?Reason=SIP%3Bcause%3D302>;index=1,<br><sip: userC@networkA;cause=480>;index=1.1 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>**CDi is performed**<br>← INVITE(Call-ID B-C)<br>**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.<br>**Check:** Is the cause parameter in the last entry is set to '480'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_cd_009 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 29 |
| **Test purpose** | **Communication Deflection immediate response, unsuccessful NDUB.**<br><br>The user A and user C are in network A. The user B is in network B.<br>Ensure that when user A calls user B, the call is deflected immediate to user C and user C is network determined user busy. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow** | |

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

| | |
|---|---|
| **Comments** | **Check:** The dialogue is terminated by receiving a 486 Busy Here.<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_cd_011 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 6.5/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55D |
| **Test purpose** | **SIP-I support. CD performed in Network B, Notification subscription options is set to presentation not allowed.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, without diverted-to user number.<br>Ensure that when user A calls user B, the call is deflected to user C, user A is not notified about call diversion.<br>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:**<br>• Calling user receives notification that his call has been diverted (forwarded or deflected) = no |
| **SIP Parameter** | 183 /180<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ACM/CPG**<br>Redirection number<br>Address signal (*Diverted-to user*)<br>Call diversion information<br>Notification subscription options<br>presentation not allowed<br>Redirecting reason<br>Deflection immediate or Deflection during alerting<br>Generic notification<br>call is diverting<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing (Call-ID B-A, ACM) in case CDa<br>**CD is performed**<br>← INVITE(Call-ID B-C, IAM)<br>← 183 / 180 (Call-ID B-A, ACM/CPG)<br>**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.<br>**Check:** Is a 183 Session Progress received at the interconnection interface?<br>**Check:** Is an ACM encapsulated in the 183?<br>**Check:** Is the Called party's status indicator set to 'no indication'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Notification subscription options indicator set to 'presentation not allowed'?<br>**Check:** Is the Redirecting reason set to 'Deflection immediate' or 'Deflection during alerting'?<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_cd_013 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 6.5/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55D |
| **Test purpose** | **SIP-I support. CD performed in Network B, Notification subscription options is set to presentation allowed with redirection number.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number.<br>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.<br>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:**<br>• Calling user receives notification that his call has been diverted (forwarded or deflected) = yes, with diverted-to user number |
| **SIP Parameter** | 183 /180<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ACM/CPG**<br>Redirection number<br>Address signal (*Diverted-to user*)<br>Call diversion information<br>Notification subscription options<br>presentation allowed with redirection number<br>Redirecting reason<br>Deflection immediate or Deflection during alerting<br>Generic notification<br>call is diverting<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing (Call-ID B-A, ACM) in case CDa<br>**CD is performed**<br>← INVITE(Call-ID B-C, IAM)<br>← 183 / 180 (Call-ID B-A, ACM/CPG)<br>**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.<br>**Check:** 183 Session Progress is received at the interconnection interface.<br>**Check:** Is an ACM encapsulated in the 183?<br>**Check:** Is the Called party's status indicator set to 'no indication'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Notification subscription options indicator is set to 'presentation allowed with redirection number'?<br>**Check:** Is the Redirecting reason set to 'Deflection immediate' or 'Deflection during alerting'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cd_014 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 6.7/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 53 AND SE 55D |
| **Test purpose** | **SIP-I support. CD performed in Network B, Restriction of the Redirection number.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Diverted-to user is subscribed to the COLR service in Permanent mode.<br>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:**<br>• Connected user subscribed to COLR, Permanent = yes |
| **SIP Parameter** | 200 OK<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ANM**<br>Redirection number restriction<br>Presentation restricted<br>--[any boundary name]-- |

**Message flow**

| **SIP (Network A)** | | **Interconnection Interface** | | **SIP (Network B)** |
|---|---|---|---|---|
| | | INVITE(Call-ID A-B) | → | |
| | ← | 180 Ringing (Call-ID B-A, ACM) in case CDa | | |
| | | **CD is performed** | | |
| | ← | INVITE(Call-ID B-C, IAM) | | |
| | | 180 Ringing (Call-ID C-B) | → | |
| | ← | 180 Ringing (Call-ID B-A, ACM/CPG) | | |
| | | 200 OK INVITE (Call-ID C-B) | → | |
| | ← | ACK (Call-ID B-C) | | |
| | ← | 200 OK INVITE (Call-ID B-A, ANM) | | |
| | | ACK (Call-ID A-B) | → | |
| | | **Apply post test routine** | | |

| | |
|---|---|
| **Comments** | Originating user in Network A establishes a call to user in Network B. Network B performs the diversion to a user in Network A.<br>**Check:** Is a 200 OK INVITE received at the interconnection interface?<br>**Check:** Is an ANM encapsulated in the 200 OK?<br>**Check:** Is the ISUP/BICC Redirection number restriction set to 'Presentation restricted'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cd_015 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 6.7/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 53 AND SE 55D |
| **Test purpose** | **SIP-I support. CD performed in Network B, No restriction of the Redirection number.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Diverted-to user is not subscribed to the COLR service.<br>Ensure that when user A calls user B, the call is deflected to user C, if a Redirection number restriction parameter is present it is set to 'Presentation allowed' in the encapsulated ANM contained in the 200 OK INVITE if ISUP/BICC- SIP-I interworking is applicable in Network A. |
| **Configuration** | **Subscription options:**<br>• Connected user subscribed to COLR = no |
| **SIP Parameter** | 200 OK<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**ANM**<br>Redirection number restriction<br>Presentation allowed<br>or<br>Redirection number restriction not present<br><br>--[any boundary name]-- |
| **Message flow** | |

| **SIP (Network A)** | | **Interconnection Interface** | | **SIP (Network B)** |
|---|---|---|---|---|
| | | INVITE(Call-ID A-B) | → | |
| | ← | 180 Ringing (Call-ID B-A, ACM) in case CDa | | |
| | | **CD is performed** | | |
| | ← | INVITE(Call-ID B-C, IAM) | | |
| | | 180 Ringing (Call-ID C-B) | → | |
| | ← | 180 Ringing (Call-ID B-A, ACM/CPG) | | |
| | | 200 OK INVITE (Call-ID C-B) | → | |
| | ← | ACK (Call-ID B-C) | | |
| | ← | 200 OK INVITE (Call-ID B-A, ANM) | | |
| | | ACK (Call-ID A-B) | → | |
| | | **Apply post test routine** | | |

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

| | |
|---|---|
| **Test case number** | SS_cd_016 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55D |
| **Test purpose** | **SIP-I support. CD performed in Network B, Notification of diverted-to user Redirecting number 'presentation allowed'.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Served user releases his/her number to diverted-to user = Release diverting number information.<br>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.<br>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:**<br>• Served user releases his/her number to diverted-to user = Release diverting number information |
| **SIP Parameter** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>Redirecting number<br>Address presentation restricted indicator<br>presentation allowed<br>Address signal (*Diverting user*)<br>Original called number<br>Address presentation restricted indicator<br>presentation allowed<br>Address signal<br>Redirection information<br>Original Redirection Reason<br>unknown<br>Redirecting indicator<br>Redirection counter<br>Redirecting reason<br>Deflection immediate or Deflection during alerting<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing (Call-ID B-A, ACM) in case CDa<br>**CD is performed**<br>← INVITE(Call-ID B-C, IAM)<br>**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.<br>**Check:** **Is an INVITE request received at the interconnection interface**?<br>**Check:** Is an IAM encapsulated in the INVITE?<br>**Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation allowed'?<br>**Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation allowed'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Redirection information present and the Redirecting reason is set to 'Deflection immediate' or 'Deflection during alerting'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cd_017 |
| **Test case group** | SIP-SIP/Service/CD |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 55D |
| **Test purpose** | **SIP-I support. CD performed in Network B, Notification of diverted-to user Redirecting number 'presentation restricted'.**<br><br>The user A and user C are in Network A. The user B is in the PSTN/PLMN part of Network B and is provided with CDi or CDa, Served user releases his/her number to diverted-to user = Release diverting number information.<br>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.<br>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:<br>• Served user releases his/her number to diverted-to user = Do not release diverting numberinformation |
| **SIP Parameter** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>Redirecting number<br>Address presentation restricted indicator<br>presentation restricted<br>Address signal (*Diverting user*)<br>Original called number<br>Address presentation restricted indicator<br>presentation restricted<br>Address signal<br>Redirection information<br>Original Redirection Reason<br>unknown<br>Redirecting indicator<br>Redirection counter<br>Redirecting reason<br>Deflection immediate or Deflection during alerting<br><br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE(Call-ID A-B) →<br>← 180 Ringing (Call-ID B-A, ACM) in case CDa<br>**CD is performed**<br>← INVITE(Call-ID B-C, IAM)<br>**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.<br>**Check:** **Is an INVITE request received at the interconnection interface**?<br>**Check:** Is an IAM encapsulated in the INVITE?<br>**Check:** Is the Redirecting number present and the Address presentation restricted indicator is set to 'presentation restricted'?<br>**Check:** Is the Original called number present and the Address presentation restricted indicator is set to 'presentation restricted'?<br>**Check:** Is the Redirection number present?<br>**Check:** Is Redirection information present and the Redirecting reason is set to 'Deflection immediate' or 'Deflection during alerting'?<br>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.**<br>The user A and user C are in network A. The user B and user D are in network B. The user B and user C are provided with CFU "Served user allows the presentation of his/her URI to diverted-to user" = No.<br><br>Ensure that when user A calls user B, the call is forwarded unconditional to user C and user C forwards to user D, and user D is not informed of the forwarding numbers. |
| **Configuration** | Subscription options:<br>• "Served user allows the presentation of his/her URI to diverted-to user" = No |
| **SIP Parameter** | INVITE:<br>History-Info:<br><sip; userB@networkB?privacy=history >;index=1,<br><sip:userC@networkA;cause=302>;index=1.1,<br><br>INVITE:<br>History-Info:<br><sip; userB@networkB?privacy=history >;index=1,<br><sip:userC@networkA;cause=302?privacy=history>;index=1.1,<br><sip:userD@networkB;cause=302>;index=1.1.1 |
| **Message flow**<br><br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br><br>INVITE(Call-ID A-B) → <br>**CFU is performed**<br>← INVITE(Call-ID B-C)<br>**CFU is performed**<br>INVITE(Call-ID C-D) →<br>**Apply post test routine** | |
| **Comments** | **Check:** Is a History-Info header containing Index number 1 present in the INVITE from User C to user D?<br>**Check:** Is a History-Info header containing Index number 1.1 present in the INVITE from User C to user D?<br>**Check:** Is a History-Info header containing Index number 1.1.1 present in the INVITE from User C to user D?<br>**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'?<br>**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'?<br>**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?<br>**Check:** Is the cause parameter in the last two entries of the History-Info Header set to '302'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_multipleCFU_002 |
| **Test case group** | SIP-SIP/Service/multibleCF |
| **Reference** | 4.5.2.6/ [9] |
| **SELECTION EXPRESSION** | SE 17 AND SE 22 |
| **Test purpose** | **Call establishment with multiple forwarding.**<br>The user A and user C are in network A. The user B and user D are in network B. The user B and user C is provided with CFU "Served user allows the presentation of his/her URI to diverted-to user" = Yes.<br><br>Ensure that when user A calls user B, the call is forwarded unconditional to user C and user C forwards to user D, and user D will be informed of the forwarding numbers. |
| **Configuration** | Subscription options:<br>• "Served user allows the presentation of his/her URI to diverted-to user" = Yes |
| **SIP Parameter** | INVITE:<br>History-Info header:<br><sip:userB@networkB>;index=1,<br><sip:userC@networkA;cause=302>;index=1.1,<br><br>INVITE:<br>History-Info header:<br><sip:userB@networkB>;index=1,<br><sip:userC@networkA;cause=302>;index=1.1,<br><sip:userD@networkB;cause=302>;index=1.1.1 |
| **Message flow**<br><br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br><br>INVITE(Call-ID A-B) →<br>**CFU is performed**<br>← INVITE(Call-ID B-C)<br>**CFU is performed**<br>INVITE(Call-ID C-D) →<br>**Apply post test routine** |
| **Comments** | **Check:** Is a History-Info header containing Index number 1 present in the INVITE from User C to user D?<br>**Check:** Is a History-Info header containing Index number 1.1.1 present in the INVITE from User C to user D?<br>**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?<br>**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?<br>**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?<br>**Check:** Is the cause parameter in the last two entries of the History-Info Header set to '302'?<br>**Check:** Is the "user=phone" parameter present in all History-Info header URIs?<br>**NOTE:** The history entries can be accumulated in "one" History-Info header or each history entry is present in one single History-Info header.<br>**NOTE:** The Request line may contain a 'cause' parameter indicating the redirecting reason.<br>Repeat this test in reverse direction. |

### 7.1.5.7 Conference (CONF)

| | |
|---|---|
| **Test case number** | SS_conf_001 |
| **Test case group** | SIP-SIP/Service/CONF |
| **Reference** | 4.5.2/ [10] |
| **SELECTION EXPRESSION** | ([Network A] SE 11 AND [Network B] SE 11) AND SE 31 |
| **Test purpose** | **3 Party establishment using the REFER method.**<br><br>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:<br>• Ensure that when user A refers to user B1 to invite to the conference, the user B1 sends a NOTIFY to user A indicating 'Trying'. The user B1 sends an INVITE request to the conference focus in network A. Is the request is confirmed, user B1 sends a NOTIFY indicating '200 OK'. User A terminates the original dialogue.<br>• Ensure that when user A refers to user B2 to invite to the conference, the user B2 sends a NOTIFY to user A indicating 'Trying'. The user B2 sends an INVITE request to the conference focus in network A. Is the request is confirmed, user B2 sends a NOTIFY indicating '200 OK'. User A terminates the original dialogue. |
| **Configuration** | |
| **SIP Parameter** | REFER(user B1)<br>Refer-To: <uri of conference focus;method=INVITE ><br><br>NOTIFY(B1, 100)<br>Content-Type: message/sipfrag<br>SIP/2.0 100<br><br>INVITE: Request URI: uri of conference focus<br>From: user B1<br><br>NOTIFY(B1, 200)<br>Content-Type: message/sipfrag<br>SIP/2.0 200 OK<br><br>REFER(user B2)<br>Refer-To: <uri of conference focus;method=INVITE ><br><br>NOTIFY(B2, 100)<br>Content-Type: message/sipfrag<br>SIP/2.0 100<br><br>INVITE: Request URI: uri of conference focus<br>From: user B2<br><br>NOTIFY(B2, 200)<br>Content-Type: message/sipfrag<br>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 was set on HOLD<br>**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'.<br>**Check:** The NOTIFY after the REFER request contains the 'SIP/2.0 100' message body.<br>**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.<br>**Check:** The NOTIFY after the REFER request contains the 'SIP/2.0 200 OK' message body.<br>**Check:** The original session is terminated by user A.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_conf_002 |
| **Test case group** | SIP-SIP/Service/CONF |
| **Reference** | 4.5.2/ [10], 4.7.2.9.7/ [20] |
| **SELECTION EXPRESSION** | [Network A] SE 12 AND SE 31 |
| **Test purpose** | **3 Party establishment using reINVITE performed by the AS in network A.**<br><br>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.<br>• Ensure that user A can invite user B1 to the conference by sending a reINVITE request.<br>• Ensure that user A can invite user B2 to the conference by sending a reINVITE request. |
| **Configuration** | |
| **SIP Parameter** | INVITE <B1><br>From: <userA><br>To: <userB1><br>Call-ID: A-B1<br>P-Asserted-Identity: <userA><br><br>SDP: a=sendrecv<br><br><br>INVITE <B2><br>From: <userA><br>Call-ID: A-B2<br>To: <userB2><br>P-Asserted-Identity: <userA><br><br>SDP: a=sendrecv |
| **Message flow** | |
| **SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>**Establish a confirmed session to user B1 from Network A to Network B and put it on hold**<br>**Establish a confirmed session to user B2 from Network A to Network B and put it on hold**<br>**User A establishes a 3PTY conversation**<br>INVITE(Call-ID A-B1) →<br>← 200 INVITE<br>ACK →<br><br>INVITE(Call-ID A-B2) →<br>← 200 INVITE<br>ACK →<br>**Apply post test routine** | |
| **Comments** | **User** A establishes a 3PTY conversation after the confirmed communication to user B1 and B2 was set on HOLD<br>**Check:** An INVITE is sent to user B1 and user B2 indicating a new IP address in the 'c' line of the SDP.<br>**Check:** The 'a' line indicates 'sendrecv'.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_conf_003 |
| **Test case group** | SIP-SIP/Service/CONF |
| **Reference** | 5.4/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 56 |
| **Test purpose** | **SIP-I/ISUP interworking. Served user establishes a 3 Party communication.**<br><br>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.<br>Ensure that when User A establishes a 3 PTY communication:<br>• an INFO request is sent to User B1 in Network B and an ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established';<br>• an INFO request is sent to User B2 in Network B and an ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established'. |
| **Configuration** | ISUP/BICC interworking applies in Network A<br>User in Network A is subscribed to the 3PTY supplementary service |
| **SIP Parameter** | INFO <B1><br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**CPG**<br>Generic Notification<br>Conference established<br><br>INFO <B2><br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**CPG**<br>Generic Notification<br>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.<br>**User A** establishes a confirmed communication to user B2 in Network B<br>**User A** invokes the 3PTY communication.<br>**Check:** Is an INFO request sent to user B1 and user B2 in Network B?<br>**Check:** Is an ISUP/BICC CPG message encapsulated in the INFO request to both remote users in Network B?<br>**Check:** Is the Generic Notification parameter in the encapsulated CPG in both INFO set to 'Conference established'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_conf_004 |
| **Test case group** | SIP-SIP/Service/CONF |
| **Reference** | 5.4/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 56 |
| **Test purpose** | **SIP-I/ISUP interworking. Served user disconnects one of the remote users.**<br><br>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.<br>Ensure that when User A disconnects the previous active user:<br>• a BYE request is sent to User B1 in Network B;<br>• 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<br>User in Network A is subscribed to the 3PTY supplementary service |
| **SIP Parameter** | INFO <B2><br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**CPG**<br>Generic Notification<br>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.<br>**User A** disconnects the communication with user B1 in Network B (previous on hold).<br>**Check:** Is a BYE request is sent to user B1 in Network B?<br>**Check:** Is an ISUP/BICC CPG message encapsulated in the INFO request to user B2 in Network B?<br>**Check:** Is the Generic Notification parameter in the encapsulated CPG in the INFO sent to user B2 set to 'Conference disconnected'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_conf_005 |
| **Test case group** | SIP-SIP/Service/CONF |
| **Reference** | 5.4/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 56 |
| **Test purpose** | **SIP-I/ISUP interworking. Served user splits the 3 Party communication.**<br><br>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:<br>• 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';<br>• 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<br>User in Network A is subscribed to the 3PTY supplementary service |
| **SIP Parameter** | INFO <B1><br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**CPG**<br>Generic Notification<br>Conference disconnected<br><br><br>INFO <B2><br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**CPG**<br>Generic Notification<br>Conference disconnected |
| **Message flow** | **SIP (Network A)** — **Interconnection Interface** — **SIP (Network B)**<br>**Establish a confirmed session from User A in Network A to user B1 in Network B and put it on hold**<br>**Establish a confirmed session from User A in Network A to user B2 in Network B**<br>**User A establishes a 3PTY conversation**<br>INFO(Call-ID A-B1, CPG) →<br>← 200 INFO<br><br>INFO(Call-ID A-B2, CPG) →<br>← 200 INFO<br>**Apply post test routine** |
| **Comments** | **User A** establishes confirmed communication to user B1 in Network B and sets it on hold.<br>**User A** establishes a confirmed communication to user B2 in Network B.<br>**Check:** Is an INFO request sent to user B1 and user B2 in Network B?<br>**Check:** Is an ISUP/BICC CPG message encapsulated in the INFO request to both remote users in Network B?<br>**Check:** Is the Generic Notification parameter in the encapsulated CPG in both INFO set to 'Conference established'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_conf_006 |
| **Test case group** | SIP-SIP/Service/CONF |
| **Reference** | 5.4/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 56 |
| **Test purpose** | **SIP-I/ISUP interworking. Establishment of aCONF conversation.**<br><br>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.<br>Ensure that when User A invokes the CONF communication:<br>• an INFO request is sent to User B1 in Network B and an ISUP/BICC CPG is encapsulated the Generic Notification is set to 'conference established' when the conference is invoked.<br><br>User A establishes a confirmed communication with a User B2 in Network B.<br>Ensure when User A adds the user B2 to the established conference:<br>• 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;<br>• 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<br>User in Network A is subscribed to the 3PTY supplementary service |
| **SIP Parameter** | INFO1 <B1><br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**CPG**<br>Generic Notification<br>conference established<br><br>INFO2 <B1><br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**CPG**<br>Generic Notification<br>Other party added<br><br>INFO3 <B2><br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**CPG**<br>Generic Notification<br>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 | |
| | INFO3(Call-ID A-B2, CPG) | → |
| ← | 200 INFO | |
| | **Apply post test routine** | |

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

| | |
|---|---|
| **Test case number** | SS_conf_007 |
| **Test case group** | SIP-SIP/Service/CONF |
| **Reference** | 5.4/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 56 |
| **Test purpose** | **SIP-I/ISUP interworking. Isolation and Reattachment of one party of the conference.**<br><br>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.<br>Ensure that when User A isolates one remote party (B1) from the CONF communication:<br>• an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'isolated' in the encapsulated ISUP/BICCCPG;<br>• 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.<br>Ensure that when User A reattaches one remote party (B1) to the CONF communication:<br>• an INFO request is sent to User B1 in Network B and the Generic Notification is set to 'reattached' in the encapsulated ISUP/BICCCPG;<br>• 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<br>User in Network A is subscribed to the 3PTY supplementary service |
| **SIP Parameter** | INFO1 <B1><br>**CPG**<br>Generic Notification= isolated<br><br>INFO2 <B1><br>**CPG**<br>Generic Notification= Other party isolated<br><br>INFO3 <B2><br>**CPG**<br>Generic Notification= reattached<br><br>INFO4 <B2><br>**CPG**<br>Generic Notification= Other party reattached |

| | |
|---|---|
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>**Establish a CONF communication with User B1 and User B2 in Network B**<br>**User A isolates User B1 from the CONF conversation**<br>INFO1(Call-ID A-B1, CPG) →<br>← 200 INFO<br>INFO3(Call-ID A-B2, CPG) →<br>← 200 INFO<br>**User A reattaches User B1 to the CONF conversation**<br>INFO2(Call-ID A-B2, CPG) →<br>← 200 INFO<br>INFO4(Call-ID A-B2, CPG) →<br>← 200 INFO<br>**Apply post test routine** | |
| **Comments** | **User A** Invokes a CONF conversation with User B1 and User b2 in Network B.<br>**User A** splits user B1 in Network B from the CONF conversation.<br>**Check:** Is an INFO request sent to user B1 and the Generic notification is set to 'isolated' in the encapsulated CPG?<br>**Check:** Is an INFO request sent to user B2 and the Generic notification is set to 'Other party isolated' in the encapsulated CPG?<br>**User A** reattaches user B1 in Network B to the CONF conversation.<br>**Check:** Is an INFO request sent to user B1 and the Generic notification is set to 'reattached' in the encapsulated CPG?<br>**Check:** Is an INFO request sent to user B2 and the Generic notification is set to 'Other party reattached' in the encapsulated CPG?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_conf_008 |
| **Test case group** | SIP-SIP/Service/CONF |
| **Reference** | 5.4/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 56 |
| **Test purpose** | **SIP-I/ISUP interworking. Splitting and Adding of a party.**<br><br>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.<br>Ensure that when User A split one remote party (B1) from the CONF communication:<br>• 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;<br>• 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.<br>Ensure that when User A adds one remote party (B1) to the CONF communication:<br>• 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;<br>• 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<br>User in Network A is subscribed to the 3PTY supplementary service |
| **SIP Parameter** | INFO1 <B1><br>**CPG**<br>Generic Notification= conference disconnected<br><br>INFO2 <B1><br>**CPG**<br>Generic Notification=Other party split<br><br>INFO3 <B2><br>**CPG**<br>Generic Notification=Conference established<br><br>INFO4 <B2><br>**CPG**<br>Generic Notification= Other party added |
| **Message flow** | |
| **SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>**Establish a CONF communication with User B1 and User B2 in Network B**<br>**User A isolates User B1 from the CONF conversation**<br>INFO1(Call-ID A-B1, CPG) →<br>← 200 INFO<br>INFO3(Call-ID A-B2, CPG) →<br>← 200 INFO<br>**User A reattaches User B1 to the CONF conversation**<br>INFO2(Call-ID A-B2, CPG) →<br>← 200 INFO<br>INFO4(Call-ID A-B2, CPG) →<br>← 200 INFO<br>**Apply post test routine** | |
| **Comments** | **User A** Invokes a CONF conversation with User B1 and User b2 in Network B.<br>**User A** splits user B1 in Network B from the CONF conversation.<br>**Check:** Is an INFO request sent to user B1 and the Generic notification is set to 'conference disconnected' in the encapsulated CPG?<br>**Check:** Is an INFO request sent to user B2 and the Generic notification is set to 'Other party split' in the encapsulated CPG?<br>**User A** adds user B1 in Network B to the CONF conversation.<br>**Check:** Is an INFO request sent to user B1 and the Generic notification is set to 'Conference established' in the encapsulated CPG?<br>**Check:** Is an INFO request sent to user B2 and the Generic notification is set to 'Other party added' in the encapsulated CPG?<br>Repeat this test in reverse direction. |

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

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|---|---|
| **Test case number** | SS_acr-cb_001 |
| **Test case group** | SIP-SIP/Service/ACR-CB |
| **Reference** | 4.5.2.6/ [12] |
| **SELECTION EXPRESSION** | SE 32 |
| **Test purpose** | **Call Barring performed in network B for user B.**<br><br>User A is located in network A and user B is located in network B and is subscribed to the Incoming Call Barring service.<br>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<br>P-Asserted-Identity: <URI of user A> |
| **Message flow** | |
| **SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 603 (Decline)<br>ACK → |
| **Comments** | **Check:** Is the P-Asserted-Identity present?<br>**Check:** Is the communication rejected by sending a 603 (Decline) final response sent to user A?<br>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/ [12] |
| **SELECTION EXPRESSION** | SE 33 |
| **Test purpose** | **ACR performed in network B for user B.**<br><br>User A is located in network A and user B is located in network B and is subscribed to the Anonymous Communication rejection service.<br>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<br>P-Asserted-Identity: <URI of user A><br>Privacy: id |
| **Message flow** | |
| **SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 433 (Anonymity Disallowed)<br>ACK → |
| **Comments** | **Check:** Is the P-Asserted-Identity present?<br>**Check:** Is the Privacy header set to 'id'?<br>**Check:** Is the communication rejected by sending a 433 (Anonymity Disallowed) final response sent to user A?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_acr-cb_003 |
| **Test case group** | SIP-SIP/Service/ACR-CB |
| **Reference** | 6.5/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND SE 47 AND SE 57 |
| **Test purpose** | **SIP-I interworking. ACR performed in network B for user B.**<br><br>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<br>P-Asserted-Identity: <URI of user A><br>Privacy: id<br><br>433<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>REL:<br>Cause indicator<br>Cause = 21 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 603 Decline (REL)<br>ACK → |
| **Comments** | **Check:** Is the P-Asserted-Identity present?<br>**Check:** Is the Privacy header set to 'id'?<br>**Check:** Is the communication rejected by sending a 603 Decline final response sent to user A?<br>**Check:** Is an ISUP/BICC REL is present in the 603 and the cause value is set to '21'?<br>Repeat this test in reverse direction. |

### 7.1.5.9 Closed User Group (CUG)

| | |
|---|---|
| **Test case number** | SS_cug_001 |
| **Test case group** | SIP-SIP/Service/CUG |
| **Reference** | 4.5.2.4/ [13] |
| **SELECTION EXPRESSION** | SE 34 |
| **Test purpose** | **Originating user +OA to terminating user no CUG.**<br><br>An originating user in a CUG Outgoing Access allowed calls a user not in a CUG. The session establishment is successful. |
| **Configuration** | **Originating user**: CUG, outgoing access allowed |
| **SIP Parameter** | INVITE:<br>Content-Type: application/vnd.etsi.cug+xml<br>Content-Disposition: signal;handling= ......<br><br>…….<br><…cug><br><…networkIndicator>01</… networkIndicator<br><…networkIndicator>23</… networkIndicator<br><….cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode><br><…CugCommunicationIndicator>10</…cugCommunicationIndicator><br><…:cug> |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>**Apply post test routine** |
| **Comments** | **Check:** Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml?<br>**Check:** Contains the XML body in the INVITE a 'cug' element?<br>**Check:** Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '10' as a 'cug' child element?<br>**Check:** Is the session setup not rejected?<br>Repeat this test in reverse direction.<br>**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/ [13] |
| **SELECTION EXPRESSION** | SE 34 |
| **Test purpose** | **Originating user -OA to terminating user no CUG.**<br><br>An originating user in a CUG Outgoing Access not allowed calls a user not in a CUG. The session establishment is not successful, a 403 (Forbidden) response is sent. |
| **Configuration** | **Originating user**: CUG, outgoing access not allowed |
| **SIP Parameter** | INVITE:<br>Content-Type: application/vnd.etsi.cug+xml<br>Content-Disposition: signal;handling= required<br><br>…….<br><…:cug><br><…networkIndicator>01</…networkIndicator<br><…networkIndicator>23</…networkIndicator<br><….ugInterlockBinaryCode>0F03</…cugInterlockBinaryCode><br><…cugCommunicationIndicator>11</…cugCommunicationIndicator><br><…cug> |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 403 (Forbidden)<br>ACK → |
| **Comments** | **Check:** Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml?<br>**Check:** Is the handling parameter in the Content-Disposition header set to required?<br>**Check:** Contains the XML body in the INVITE a 'cug' element?<br>**Check:** Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element?<br>**Check:** Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network?<br>Repeat this test in reverse direction.<br>**NOTE:** The networkIndicator element value and the cugInterlockBinaryCode element value are examples. |

| | |
|---|---|
| **Test case number** | SS_cug_003 |
| **Test case group** | SIP-SIP/Service/CUG |
| **Reference** | 4.5.2.4, 4.5.2.10/ [13] |
| **SELECTION EXPRESSION** | SE 34 |
| **Test purpose** | **Originating user -OA to terminating user -IA.**<br><br>An originating user in a CUG Outgoing Access not allowed calls a user in the same CUG Incoming Access not allowed. The session establishment is successful. |
| **Configuration** | **Originating user**: CUG, outgoing access not allowed<br>**Terminating user**: CUG incoming access not allowed<br>User in network A and user in network B are in the same CUG |
| **SIP Parameter** | INVITE:<br>Content-Type: application/vnd.etsi.cug+xml<br>Content-Disposition: signal;handling= required<br><br>…….<br><…cug><br><…networkIndicator>01</…networkIndicator<br><…networkIndicator>23</…networkIndicator<br><…cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode><br><…cugCommunicationIndicator>11</…cugCommunicationIndicator><br><…cug> |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>**Apply post test routine** |
| **Comments** | **Check:** Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml?<br>**Check:** Is the handling parameter in the Content-Disposition header set to required?<br>**Check:** Contains the XML body in the INVITE a 'cug' element?<br>**Check:** Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element?<br>**Check:** Is the session setup not rejected?<br>Repeat this test in reverse direction.<br>**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/ [13] |
| **SELECTION EXPRESSION** | SE 34 |
| **Test purpose** | **Originating user in a CUG to terminating user -IA.**<br><br>An originating user in a CUG calls 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<br>**Terminating user:** CUG incoming access not allowed |
| **SIP Parameter** | INVITE:<br>Content-Type: application/vnd.etsi.cug+xml<br>Content-Disposition: signal;handling= .....<br><br>…….<br><…cug><br><…networkIndicator>01</…networkIndicator<br><…networkIndicator>23</…networkIndicator<br><…cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode><br><…cugCommunicationIndicator>..</…cugCommunicationIndicator><br><…cug> |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 403 (Forbidden)<br>ACK → |
| **Comments** | **Check:** Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml?<br>**Check:** Contains the XML body in the INVITE a 'cug' element?<br>**Check:** Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '10' or '11'as a 'cug' child element?<br>**Check:** Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network?<br>Repeat this test in reverse direction.<br>**NOTE:** The networkIndicator element value and the cugInterlockBinaryCode element value are examples. |

| | |
|---|---|
| **Test case number** | SS_cug_005 |
| **Test case group** | SIP-SIP/Service/CUG |
| **Reference** | 4.5.2.10/ [13] |
| **SELECTION EXPRESSION** | SE 34 |
| **Test purpose** | **Originating user no CUG to terminating user +IA.**<br><br>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**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>**Apply post test routine** |
| **Comments** | **Check:** Is the session setup not rejected?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cug_006 |
| **Test case group** | SIP-SIP/Service/CUG |
| **Reference** | 4.5.2.10/ [13] |
| **SELECTION EXPRESSION** | [Network A] SE 34 AND NOT [Network B] SE 34 |
| **Test purpose** | **Originating user no CUG to terminating user -IA.**<br><br>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**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 403 (Forbidden)<br>ACK → |
| **Comments** | **Check:** Is the session setup rejected? A 403 (Forbidden) final response is sent by the terminating network.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cug_007 |
| **Test case group** | SIP-SIP/Service/CUG |
| **Reference** | 4.5.2.4/ [13] |
| **SELECTION EXPRESSION** | SE 34 |
| **Test purpose** | **Originating user -OA, network B does not support CUG.**<br><br>An originating user in a CUG Outgoing Access not allowed calls 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:<br>Content-Type: application/vnd.etsi.cug+xml<br>Content-Disposition: signal;handling= required<br><br>…….<br><…cug><br><…networkIndicator>01</…networkIndicator<br><…networkIndicator>23</…networkIndicator<br><…cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode><br><…cugCommunicationIndicator>11</…cugCommunicationIndicator><br><…cug> |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 4xx/501 Not Implemented<br>ACK → |
| **Comments** | **Check:** Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml?<br>**Check:** Is the handling parameter in the Content-Disposition header set to required?<br>**Check:** Contains the XML body in the INVITE a 'cug' element?<br>**Check:** Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element?<br>**Check:** Is the session setup rejected by sending an unsuccessful final response?<br>Repeat this test in reverse direction.<br>**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/ [13] |
| **SELECTION EXPRESSION** | SE 34 |
| **Test purpose** | **Originating user CUG-OA to terminating CUG user +ICB**<br><br>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:<br>Content-Type: application/vnd.etsi.cug+xml<br>Content-Disposition: signal;handling= required<br><br>…….<br><…cug><br><…networkIndicator>01</…networkIndicator<br><…networkIndicator>23</…networkIndicator<br><…cugInterlockBinaryCode>0F03</…cugInterlockBinaryCode><br><…cugCommunicationIndicator>11</…cugCommunicationIndicator><br><…cug> |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 603 Decline<br>ACK → |
| **Comments** | **Check:** Is the Content-Type in The INVITE set to application/vnd.etsi.cug+xml?<br>**Check:** Is the handling parameter in the Content-Disposition header set to required?<br>**Check:** Contains the XML body in the INVITE a 'cug' element?<br>**Check:** Contains the XML body in the INVITE a 'networkIndicator' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugInterlockBinaryCode' element as a 'cug' child element?<br>**Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element?<br>**Check:** Is the session setup rejected by sending a **603 Decline** unsuccessful final response?<br>Repeat this test in reverse direction.<br>**NOTE:** The networkIndicator element value and the cugInterlockBinaryCode element value are examples. |

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

| | |
|---|---|
| **Test case number** | SS_cug_009 |
| **Test case group** | SIP-SIP/Service/CUG |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 58 |
| **Test purpose** | **SIP-I/ISUP interworking. CUG call with outgoing access not allowed (both user in different CUG).**<br><br>User A is located in the PSTN part of Network A and ISUP/BICC interworking applies in Network A. ensure that when user A is in a CUG 'outgoing access allowed' calls user B in Network B. The call is unsuccessful due to both CUG user are in different CUG. There is an Optional forward call indicator the CUG Call Indicator **Outgoing access not allowed** present in the encapsulated IAM sent to Network B. |
| **Configuration** | • User in PSTN/PLMN part of Network A in a CUG outgoing access not allowed |
| **SIP Parameter** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>**Optional Forward call indicator**<br>CUG Call Indicator<br>Outgoing access not allowed<br>**CUG interlock code**<br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 500 Server Internal error<br>ACK →<br>**Apply post test routine** |
| **Comments** | **User A** in the PSTN part of Network A calls user B in Network B.<br>**Check:** Is an IAM encapsulated in the INVITE request sent from Network A to Network B?<br>**Check:** Is the Optional forward call indicator present, the CUG Call Indicator is set to 'Outgoing access not allowed'?<br>**Check:** Is the CUG interlock code parameter present in the encapsulated IAM?<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_cug_011 |
| **Test case group** | SIP-SIP/Service/CUG |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | ([Network A] SE 17 AND [Network A] SE 47 AND SE 58) AND ([Network B] SE 17 AND [Network B] SE 47 AND SE 58) |
| **Test purpose** | **SIP-I/ISUP interworking. CUG call to a CUG user incoming access not allowed (both user in the same CUG).**<br><br>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<br>• User in PSTN/PLMN part of Network B in a CUG incoming access not allowed<br>• User A and User B are in the same CUG |
| **SIP Parameter** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>**Optional Forward call indicator**<br>CUG Call Indicator<br>Outgoing access not allowed<br>**CUG interlock code**<br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>**Apply post test routine** |
| **Comments** | **User A** in the PSTN/PLMN part of Network A calls user B in Network B.<br>**User B** in the PSTN/PLMN part of Network B.<br>**Check:** Is an IAM encapsulated in the INVITE request sent from Network A to Network B?<br>**Check:** Is the Optional forward call indicator present, the CUG Call Indicator is set to 'Outgoing access not allowed'?<br>**Check:** Is the CUG interlock code parameter present in the encapsulated IAM?<br>**Check:** Is the call setup successful?<br>Repeat this test in reverse direction. |

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

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

| | |
|---|---|
| **Test case number** | SS_cug_014 |
| **Test case group** | SIP-SIP/Service/CUG |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND SE 47 AND SE 58 |
| **Test purpose** | **SIP-I/ISUP interworking. Call to a CUG user incoming access allowed.**<br><br>User A is located in Network A and not in a CUG. User B is located in the PSTN/PLMN part and SIP-I - ISUP/BICC interworking applied. Ensure that when user A calls CUG user B Incoming access allowed in Network B. The call is successful. |
| **Configuration** | • User in PSTN/PLMN part of Network B in a CUG incoming access allowed |
| **SIP Parameter** | |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>**Apply post test routine** |
| **Comments** | **User A** in Network A calls user B in Network B<br>**User B** in the PSTN/PLMN part of Network B.<br>**Check:** Is the call setup successful?<br>Repeat this test in reverse direction. |

### 7.1.5.10 Communication Waiting (CW)

| | |
|---|---|
| **Test case number** | SS_cw_001 |
| **Test case group** | SIP-SIP/Service/CW |
| **Reference** | 4.5.5.2/ [15] |
| **SELECTION EXPRESSION** | SE 35 |
| **Test purpose** | **Call Waiting indication in 180 response.**<br><br>User A is located in network A, user B is located in network B and subscribed to the communication Waiting service.<br>Ensure that when user A calls user B, user A receives the 'communication Waiting indication' in the 180 Ringing provisional response if the user B is NDUB or UDUB. |
| **Configuration** | User B subscribed to the CW service |
| **SIP Parameter** | 180:<br>Alert-Info: <urn:alert:service:call-waiting> |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>**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>'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cw_002 |
| **Test case group** | SIP-SIP/Service/CW |
| **Reference** | 4.5.5.2/ [15] |
| **SELECTION EXPRESSION** | SE 35 AND SE 36 |
| **Test purpose** | **Call rejected after timeout TAS-CW.**<br><br>User A is located in network A, user B is located in network B and subscribed to the communication Waiting service.<br>Ensure that when user A calls user B, user A receives the 'communication Waiting indication' in the 180 Ringing provisional response if the user B is NDUB or UDUB. After timeout TAS-CW network B sends a 480 (Temporarily unavailable) response toward user A and the Reason header field is set to '19'. |
| **Configuration** | |
| **SIP Parameter** | 180:<br>Alert-Info: <urn:alert:service:call-waiting><br><br>480:<br>Reason: Q.850 ;cause=19 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>Timeout TAS-CW<br>← 480 (Temporarily unavailable)<br>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>'?<br>**Check:** Is a Reason header present in the 480 Response and is the protocol is set to 'Q.850' and the cause parameter set to '19'?<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_cw_004 |
| **Test case group** | SIP-SIP/Service/CW |
| **Reference** | 6.5/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND [Network B] SE 47 AND SE 59 |
| **Test purpose** | **SIP-I support. Call Waiting indication in 180 with encapsulated CPG.**<br><br>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.<br>Ensure that when user A calls user B, an encapsulated ISUP/BICC CPG Generic notification 'call is a waiting call' is present in the 180 Ringing provisional response if the user B is NDUB. |
| **Configuration** | User B subscribed to the CW service |
| **SIP Parameter** | 180<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**CPG**<br>**Event information**<br>Event indicator<br>ALERTING<br>**Generic notification**<br>Notification indicator<br>call is a waiting call |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 183 Session Progress (ACM)<br>← 180 Ringing (CPG)<br>**Apply post test routine** |
| **Comments** | **Check:** Is an ISUP/BICC CPG present in the 180 provisional response and the Generic notification is set to 'call is a waiting call'?<br>Repeat this test in reverse direction. |

### 7.1.5.11 Explicit Communication Transfer (ECT)

| | |
|---|---|
| **Test case number** | SS_ect_001 |
| **Test case group** | SIP-SIP/Service/ECT |
| **Reference** | 4.5.2/ [11] |
| **SELECTION EXPRESSION** | [Network A] SE 37 AND [Network A] SE 11 AND [Network A] SE 49 |
| **Test purpose** | **Blind/assured transfer using the REFER method.**<br><br>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.<br>• 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'.<br>• 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.<br>• 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<br>Refer-To: <URI of ECT-AS>; method=invite<br><br>INVITE1 Request URI address of ECT-AS<br><br>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 is sent network B, the Refer-To header is set to the URI of the ECT-AS in network A and a method parameter is present set to 'INVITE'?<br>**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?<br>**Check:** Is an INVITE request sent to network A the Request line is set to the address of the ECT-AS in network A?<br>**Check:** Is an INVITE request is sent to network B the Request is set to the address of user C?<br>**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.<br>**Check:** Ensure the property of speech between user B and user C.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_ect_002 |
| **Test case group** | SIP-SIP/Service/ECT |
| **Reference** | 4.5.2/ [11] |
| **SELECTION EXPRESSION** | [Network A] SE 37 AND [Network A] SE 11 AND [Network A] SE 50 |
| **Test purpose** | **Consultative transfer using the REFER method.**<br><br>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.<br>• 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'.<br>• 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.<br>• 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<br>Refer-To: <URI of ECT-AS>; method=invite<br><br>INVITE1 Request URI address of ECT-AS<br><br>INVITE2: Request URI address of user C<br>Require: replaces<br>Replaces: <session A-C> |
| **Message flow** | **SIP (Network A) — Interconnection Interface — SIP (Network B)**<br>**A confirmed session is established between user A and user B**<br>**A confirmed session is established between user A and user C**<br>**User A invokes ECT to transfer the session to user C**<br>REFER →<br>← 202 Accepted<br>← NOTIFY (100)<br>200 OK NOTIFY →<br><br>← INVITE1 (ECT-AS)<br>INVITE2 (user C) →<br>← 200 OK INVITE<br>ACK →<br>200 OK INVITE →<br>← ACK<br>← NOTIFY (200)<br>200 OK NOTIFY →<br><br>BYE (A-B) →<br>← 200 OK BYE<br><br>← BYE (A-C)<br>200 OK BYE →<br>**Apply post test routine** |
| **Comments** | **Check:** Is a REFER request is sent network B, the Refer-To header is set to the URI of the ECT-AS in network A and a method parameter is present set to 'INVITE'?<br>**Check:** Is an INVITE request sent to network A the Request line is set to the address of the ECT-AS in network A?<br>**Check:** Is an INVITE request is sent to network B the Request is set to the address of user C and a Replaces header is present contains the session identifiers of the session A-C?<br>**Check:** Is the session A - B and the session A - C terminated?<br>**Check:** Ensure the property of speech between user B and user C.<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_ect_004 |
| **Test case group** | SIP-SIP/Service/ECT |
| **Reference** | 4.5.2/ [11], 4.7.2.9.7/ [20] |
| **SELECTION EXPRESSION** | [Network A] SE37 AND [Network A] SE 12 AND [Network A] SE 50 |
| **Test purpose** | **Consultative transfer using the 3pcc method.**<br><br>User A is located in network A, user B and user C are located in network B<br>User A invokes ECT to transfer a session with user B to user C.<br>• Ensure that the network A sends a reINVITE to update the session between user A and user B (SDP: IP address, port and codec).<br>• Ensure that the network A sends a reINVITE to update the session between user A and user C (SDP: IP address, port and codec). |
| **Configuration** | |
| **SIP Parameter** | INVITE1: Request URI address of user C<br>SDP<br>c=IN IP4/6 [new IP address]<br>m=audio [new port] RTP/AVP [new codec list]<br>a=[new attributes]<br><br>**CASE A**<br>INVITE2: Request URI address of user B<br>SDP<br>c=IN IP4/6 [new IP address]<br>m=audio [new port] RTP/AVP [new codec list]<br>a=[new attributes]<br><br>**CASE B**<br>INVITE2: Request URI address of user B<br>SDP<br>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?<br>**Check:** Is a reINVITE is sent from network A to user C update the session parameter in the SDP? CASE A reflects the ECT procedure in the IMS, CASE B reflects the interworking in the MGCF.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_ect_005 |
| **Test case group** | SIP-SIP/Service/ECT |
| **Reference** | 5.4.3.2/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 60 |
| **Test purpose** | **SIP-I support. Call Transfer invoked in active state, call was previous on HOLD.**<br><br>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.<br>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<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/sdp<br>a=sendrecv<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**FAC**<br>Generic Notification<br>Call transfer active<br>Call transfer number<br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>**A confirmed session is established between user A and user B and set on hold**<br>**User A invokes ECT to transfer the session to user C**<br>*INFO (LOP request)* →<br>← *200 OK INFO*<br>← *INFO (LOP response)*<br>*200 OK INFO* →<br>**CASE A**<br>INVITE (sendrecv; **FAC**) →<br>← 200 OK INVITE<br>ACK →<br>**CASE B**<br>INFO (**FAC**) →<br>← 200 OK INFO<br>INVITE (sendrecv) →<br>← 200 OK INVITE<br>ACK →<br>**Apply post test routine** | |
| **Comments** | A session from User A to User B is already established.<br>User A sets the User B on hold.<br>User A invokes the ECT service.<br>**Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**request**'?<br>**Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**response**'?<br>**Check:** Is (CASE A) an INVITE request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer active' and in addition the media stream is set to 'sendrecv'?<br>**Check:** Is (CASE B) an INFO request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer active'? In addition is an **INVITE** request sent and the media stream is set to 'sendrecv' to resume the held session?<br>**NOTE:** The content of the FAC in the INVITE request is Equal to the content of the FAC in the INFO request.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_ect_006 |
| **Test case group** | SIP-SIP/Service/ECT |
| **Reference** | 5.4.3.2/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 60 |
| **Test purpose** | **SIP-I support. Call Transfer invoked in alerting state, call was previous on HOLD.**<br><br>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.<br>Ensure that a 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** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/sdp<br>a=sendrecv<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**FAC**<br>Generic Notification<br>Call transfer alerting<br>Call transfer number<br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A) Interconnection Interface SIP (Network B)**<br>**A confirmed session is established between user A and user B and set on hold**<br>**User A invokes ECT to transfer the session to user C**<br>*INFO (LOP request)* →<br>← *200 OK INFO*<br>← *INFO (LOP response)*<br>*200 OK INFO* →<br>**CASE A**<br>INVITE (sendrecv; **FAC**) →<br>← 200 OK INVITE<br>ACK →<br>**CASE B**<br>INFO (**FAC**) →<br>← 200 OK INFO<br>INVITE (sendrecv) →<br>← 200 OK INVITE<br>ACK →<br>**Apply post test routine** | |
| **Comments** | A session from User A to User B is already established.<br>User A sets the User B on hold.<br>A session from User A to User C is already established.<br>User A invokes the ECT service.<br>**Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**request**'?<br>**Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**response**'?<br>**Check:** Is (CASE A) an INVITE request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer alerting' and in addition the media stream is set to 'sendrecv'?<br>**Check:** Is (CASE B) an INFO request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer alerting'? In addition is an **INVITE** request sent and the media stream is set to 'sendrecv' to resume the held session?<br>**NOTE:** The content of the FAC in the INVITE request is Equal to the content of the FAC in the INFO request.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_ect_007 |
| **Test case group** | SIP-SIP/Service/ECT |
| **Reference** | 5.4.3.2/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 60 |
| **Test purpose** | **SIP-I support. Call Transfer invoked in active state.**<br><br>BICC/ISUP - SIP-I interworking applies in the originating network Users A and B are located in network A and User C is located in network B.<br>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** | INFO<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**FAC**<br>Generic Notification<br>Call transfer active<br>Call transfer number |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>**A confirmed session is established between user A and user C**<br>**User A invokes ECT to transfer the session to user C**<br>*INFO (LOP request)* →<br>← *200 OK INFO*<br>← *INFO (LOP response)*<br>*200 OK INFO* →<br><br>INFO (**FAC**) →<br>← 200 OK INFO<br>**Apply post test routine** | |
| **Comments** | A session from User A to User B is already established.<br>User A sets the User B on hold.<br>A session from User A to User C is already established.<br>User A invokes the ECT service.<br>**Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**request**'?<br>**Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**response**'?<br>**Check:** Is (CASE B) an INFO request sent and an ISUP FAC message is present containing a Generic notification indicator is set to 'Call transfer active'?<br>**NOTE:** The content of the FAC in the INVITE request is Equal to the content of the FAC in the INFO request.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_ect_008 |
| **Test case group** | SIP-SIP/Service/ECT |
| **Reference** | 5.4.3.2/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 60 |
| **Test purpose** | **SIP-I support. Call Transfer invoked in alerting state.**<br><br>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.<br>Ensure that a 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<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**CPG**<br>Generic Notification<br>Call transfer alerting<br>Call transfer number |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>**A session in the early dialogue is established between user A and user C**<br>**User A invokes ECT to transfer the session to user C**<br>*INFO (LOP request)* →<br>← *200 OK INFO*<br>← *INFO (LOP response)*<br>*200 OK INFO* →<br><br>INFO (**CPG**) →<br>← 200 OK INFO<br>**Apply post test routine** | |
| **Comments** | A session from User A to User B is already established.<br>User A sets the User B on hold.<br>A session from User A to User C is already established.<br>User A invokes the ECT service.<br>**Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**request**'?<br>**Check:** Is (**optional**) an INFO request is sent from Network A to Network B and an ISUP LOP message is present the Loop prevention indicator set to '**response**'?<br>**Check:** Is (CASE B) an INFO request sent and an ISUP CPG message is present containing a Generic notification indicator is set to 'Call transfer alerting'?<br>**NOTE:** The content of the FAC in the INVITE request is Equal to the content of the FAC in the INFO request.<br>Repeat this test in reverse direction. |

### 7.1.5.12 Malicious Communication Identification (MCID)

<table>
<tr><td>Test case number</td><td>SS_mcid_001</td></tr>
<tr><td>Test case group</td><td>SIP-SIP/Service/MCID</td></tr>
<tr><td>Reference</td><td>4.5.2.5/ [18]</td></tr>
<tr><td>SELECTION EXPRESSION</td><td>SE 38</td></tr>
<tr><td>Test purpose</td><td>Network B sends a MCID request, no response.<br><br>User A is located in network A, user B is located in network B and subscribed to the Malicious Communication Identification service.<br>When user A call user B and no originating identification is present in the INVITE request, the network B sends an INFO request to network A requesting the originating identity. After timeout of timer TO-ID the network B sends the 180 Ringing response.</td></tr>
<tr><td>Configuration</td><td>User B is subscribed to the MCID service</td></tr>
<tr><td>SIP Parameter</td><td>INFO:<br><…:mcid……………...><br><…:request><br><…:McidRequestIndicator>01</…:McidRequestIndicator><br><…:HoldingIndicator >…</…:HoldingIndicator><br></…:request><br></…:mcid></td></tr>
<tr><td colspan="2">Message flow<br>SIP (Network A) | Interconnection Interface | SIP (Network B)<br>INVITE →<br>← INFO<br>200 OK INFO →<br>Timeout $T_{O-ID}$<br>← 180 Ringing<br>Apply post test routine</td></tr>
<tr><td>Comments</td><td>Check: Is an INFO request sent to network A?<br>Check: Is the McidRequestIndicator element set to ,01'?<br>Check: is a 200 OK INFO response sent to network B?<br>Repeat this test in reverse direction.</td></tr>
</table>

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

| | |
|---|---|
| **Test case number** | SS_mcid_003 |
| **Test case group** | SIP-SIP/Service/MCID |
| **Reference** | 5.4.3.2/ [24] |
| **SELECTION EXPRESSION** | ([Network B] SE 17 AND SE 47 AND SE 61) |
| **Test purpose** | **SIP-I support. Network B sends a MCID request, no response.**<br><br>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.<br>When user A call user B and no originating identification is present in the INVITE request, the network B sends a 183 Session Progress to network A and an ISUP/BICC IDR message is present the MCID request indicator is set to 'MCID requested' requesting the originating identity. After timeout of timer (ISUP) T39 the network B sends the 180 Ringing response. |
| **Configuration** | User B is subscribed to the MCID service |
| **SIP Parameter** | 183:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IDR**<br>**MCID request indicators**<br>MCID request indicator<br>MCID requested |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 183 Session Progress(IDR)<br><br>**Timeout T39**<br>← 180 Ringing(ACM)<br>**Apply post test routine** |
| **Comments** | **Check:** Is a 183 Session Progress sent to network A?<br>**Check:** Is an ISUP/BICC IDR message is present and the MCID request indicator is set to 'MCID requested'?<br>**NOTE:** Based on network policies the MCID request indicator can be set to 'MCID not requested'.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_mcid_004 |
| **Test case group** | SIP-SIP/Service/MCID |
| **Reference** | 5.4.3.2/ [24] |
| **SELECTION EXPRESSION** | ([Network A] SE 17 AND SE 47 AND SE 61) AND ([Network B] SE 17 AND SE 47 AND SE 61) |
| **Test purpose** | **SIP-I support. Network B sends a MCID request, MCID response.**<br><br>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.<br>When user A call user B and no originating identification is present in the INVITE request, the network B sends a 183 Session Progress to network A requesting the originating identity. After receipt of an INFO request containing the ISUP IRS message from network A the network B sends the 180 Ringing response. |
| **Configuration** | User B subscribed to the MCID service<br>User A is a ISDN or POTS user in the PSTN of network A |
| **SIP Parameter** | 183:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IDR**<br>**MCID request indicators**<br>MCID request indicator<br>MCID requested<br><br>INFO:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IRS**<br>**MCID response indicators**<br>MCID response indicator<br>MCID included<br>**Calling party number** |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE ➔<br>🡐 183 Session Progress(IDR)<br>INFO(IRS) ➔<br>🡐 200 OK INFO<br>🡐 180 Ringing(ACM)<br>**Apply post test routine** |
| **Comments** | **Check:** Is a 183 Session Progress sent to network A and an ISUP/BICC IDR is present and the MCID request indicator is set to 'MCID requested'?<br>**Check:** Is an INFO request sent to network B and an ISUP/BICC IRS is present and the MCID response indicator is set to 'MCID included'?<br>**Check:** Is the Calling party number present in the attached ISUP/BICC IRS?<br>**Check:** Is a 200 OK INFO response sent to network A?<br>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/ [16] |
| **SELECTION EXPRESSION** | [Network A] SE 39 AND [Network B] SE 39 |
| **Test purpose** | **Initial subscription of a Voicemail box.**<br><br>The Voicemail owner is in network A, his Voicemail box is located in network B.<br>Ensure that a Voicemail owner is able to activate his Voicemail box. |
| **Configuration** | Voicemail in network B<br>Voicemail owner in network A |
| **SIP Parameter** | SUBCRIBE<br>Event: message-summary<br>Expires: [any value]<br>Accept: application/simple-message-summary<br><br>NOTIFY<br>Subscription-State: active;expires=[any value]<br>Event: message-summary |
| **Message flow** | |
| **SIP (Network A)** | **Interconnection Interface** \| **SIP (Network B)**<br>SUBCRIBE →<br>← 200 OK SUBSCRIBE<br><br>← NOTIFY<br>200 OK NOTIFY →<br>← 200 OK BYE<br><br>← NOTIFY<br>200 OK NOTIFY →<br>**Apply post test routine** |
| **Comments** | **Check:** Is it possible for a user in network A to subscribe to a Voicemail box in network B?<br>**Check:** Is the Event header in the SUBCRIBE set to 'message-summary'?<br>**Check:** Is the Accept header in the SUBCRIBE set to 'application/simple-message-summary'?<br>**Check:** Is the Event header in the NOTIFY is set to 'message-summary'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_mwi_002 |
| **Test case group** | SIP-SIP/Service/MWI |
| **Reference** | 4.7.2/ [16] |
| **SELECTION EXPRESSION** | [Network A] SE 39 AND [Network B] SE 39 |
| **Test purpose** | **A new entry in the Voicemail box is indicated to the owner.**<br><br>The Voicemail owner is in network A, his Voicemail box is located in network B. Ensure when a user calls user A and the call is not answered, the call is forwarded to the Voicemail box of user A in network B. Ensure that the user A is notified by message waiting indication that there is a new message present in his voicemail account. |
| **Configuration** | Voicemail in network B<br>Voicemail owner in network A |
| **SIP Parameter** | NOTIFY<br>Subscription-State: active;expires=[any value]<br>Event: message-summary<br>Content-Type: application/simple-message-summary<br><br>Messages-Waiting: yes<br>Message-Account: sip:userA@networkA (optional)<br>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'?<br>**Check:** Is the Content-Type header in the NOTIFY set to 'application/simple-message-summary'?<br>**Check:** Contains the MIME body the header 'Messages-Waiting' set to 'yes'?<br>**Check:** Contains the MIME body the optional header 'Message-Account'?<br>**Check:** Contains the MIME body the optional header 'Voice-Message'?<br>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/ [14] |
| **SELECTION EXPRESSION** | [Network B] SE 40 |
| **Test purpose** | **Indicating of CCBS possible.**<br><br>User A is located in network A and user B is located in network B.<br>Ensure when user A calls user B and user B is busy, the network B sends an indication that CCBS is possible in the 486 Busy Here final response. |
| **Configuration** | |
| **SIP Parameter** | 486:<br>Call-Info: <sip:UE-B>;purpose=call-completion;m=BS |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 486 Busy Here<br>ACK → |
| **Comments** | **Check:** The 486 final response contains the Call-Info header.<br>**Check:** The Call-Info header contains the URI of user B as the monitor point in network B.<br>**Check:** The Call-Info header contains the purpose parameter set to 'call-completion' and the m parameter set to 'BS'.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cc_002 |
| **Test case group** | SIP-SIP/Service/CC |
| **Reference** | 4.5.4.3/ [14] |
| **SELECTION EXPRESSION** | [Network B] SE 41 |
| **Test purpose** | **Indicating of CCNR possible.**<br><br>User A is located in network A and user B is located in network B.<br>Ensure when user A calls user B and user B is free, the network B sends an indication that CCNR is possible in the 180 Ringing provisional response. |
| **Configuration** | |
| **SIP Parameter** | 180:<br>Call-Info: <sip:UE-B>;purpose=call-completion;m=NR |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>**Apply post test routine** |
| **Comments** | **Check:** The 180 provisional response contains the Call-Info header.<br>**Check:** The Call-Info header contains the URI of user B as the monitor point in network B.<br>**Check:** The Call-Info header contains the purpose parameter set to 'call-completion' and the m parameter set to 'NR'.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_cc_003 |
| **Test case group** | SIP-SIP/Service/CC |
| **Reference** | 4.5.4.2/ [14] |
| **SELECTION EXPRESSION** | ([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR [Network B] SE 41) |
| **Test purpose** | **Invocation of CCBS or CCNR.**<br><br>User A is located in network A and user B is located in network B.<br>• Ensure when user A call user B and user B is busy, the indication that CCBS is possible is sent to the network A. when user A invokes CCBS, a SUBSCRIBE request is sent to the network B, the Event header is set to 'call-completion' and the m parameter in the Request line is set to 'BS'.<br>• Ensure when user A call user B and user B is free, the indication that CCNR is possible is sent to the network A. when user A invokes CCNR, a SUBSCRIBE request is sent to the network B, the Event header is set to 'call-completion' and the m parameter in the Request line is set to 'NR'.<br>Ensure that the network B sends a NOTIFY request to network A to confirm that the request is in the Call completion queue at the terminating Application Server. |
| **Configuration** | |
| **SIP Parameter** | SUBSRIBE sip:B-AS;m=**BS** or m=**NR**<br>From:<UE-A><br>To:<UE-B><br>Contact:<A-AS><br>Event:call-completion<br><br>NOTIFY sip:A-AS<br>Event:call-completion<br>Content-Type: application/call-completion<br>state: queued<br>service-retention |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>**An indication whether CCBS or CCNR is possible is sent by network B**<br>SUBSCRIBE →<br>← 202 Accepted<br><br>← NOTIFY<br>200 OK NOTIFY →<br>**Apply post test routine** | |
| **Comments** | **Check:** Is a SUBCRIBE request is sent to network B?<br>**Check:** Is the m parameter in the Request URI is set to '**BS'** in case of CCBS request or set to '**NR'** in case of CCNR?<br>**Check:** Is a NOTIFY request is sent to network A and the Event header is set to 'call-completion' and the state header in the message body is set to 'queued''?<br>Repeat this test in reverse direction.<br>**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/ [14] |
| **SELECTION EXPRESSION** | ([Network A] SE 40 OR [Network A] SE 41) AND ([Network B] SE 40 OR [Network B] SE 41) |
| **Test purpose** | **Invocation of CCBS or CCNR unsuccessful; short term denial**<br><br>User A is located in network A and user B is located in network B.<br><br>Ensure that user A invokes a CCBS or CCNR request to network B and the network B is currently unable to process the request (e.g. the B-queue is full), a 480 Temporarily Unavailable final response is sent. |
| **Configuration** | |
| **SIP Parameter** | SUBSRIBE sip:B-AS;m=BS or m=NR<br>From:<UE-A><br>To:<UE-B><br>Contact:<A-AS><br>Event:call-completion |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>**An indication whether CCBS or CCNR is possible is sent by network B**<br>SUBSCRIBE →<br>← 480 (Temporarily Unavailable) | |
| **Comments** | **Check:** Is a SUBCRIBE request is sent to network B?<br>**Check:** Is the m parameter in the Request URI is set to 'BS' in case of CCBS request or set to 'NR' in case of CCNR?<br>**Check:** Is a 480 Temporarily Unavailable sent from network B indicates the CCBS or CCNR request is unsuccessful e.g. CC queue is full?<br>Repeat this test in reverse direction. |

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

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

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

| | |
|---|---|
| **Test case number** | SS_cc_008 |
| **Test case group** | SIP-SIP/Service/CC |
| **Reference** | 6.11.2/ [24] |
| **SELECTION EXPRESSION** | [Network B] SE 17 AND SE 47 |
| **Test purpose** | **SIP-I support: Indicating of CCBS possible.**<br><br>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.<br>Ensure when user A calls user B and user B is busy, the network B sends a 486 Busy Here final response and an encapsulated ISUP REL is present, the Cause value indicator is set to #17 or #34 and the CCBS possible indicator is set to 'CCBS possible'. |
| **Configuration** | |
| **SIP Parameter** | 486:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**REL**<br>Cause value<br>#17 or #34<br>Diagnostics<br>CCBS possible |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 486 Busy Here (**REL**)<br>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'.<br>Repeat this test in reverse direction. |

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

## 7.1.6 Other PSTN services (SIP-I interworking)

### 7.1.6.1 User-to-User Signalling (UUS)

| | |
|---|---|
| **Test case number** | SS_uus_001 |
| **Test case group** | SIP-SIP/SIP-I/UUS |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 63 |
| **Test purpose** | **SIP-I support: Indicating of User-to-User service 1 implicit in initial INVITE request.**<br><br>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.<br>Ensure when user A subscribed to the User-to-User service 1 implicit request calls user B and a **User-to-user Information** parameter is present in the encapsulated IAM of the initial INVTE request. |
| **Configuration** | User A is subscribed to the User-to-User service 1 implicit request |
| **SIP Parameter** | INVITE:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br><br>**IAM**<br>User-to-user Information<br>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?<br>**Check:** Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_uus_002 |
| **Test case group** | SIP-SIP/SIP-I/UUS |
| **Reference** | 7.1, 6.5/ [24] |
| **SELECTION EXPRESSION** | ([Network A] SE 17 AND [Network A] SE 47) AND ([Network B] SE 17 AND [Network B] SE 47) AND SE 63 |
| **Test purpose** | **SIP-I support: Indicating of User-to-User service 1 implicit response in 180 or 200 OK.**<br><br>BICC/ISUP - SIP-I interworking applies in the originating and terminating network<br>User A is located in network A and user B is located in network B.<br>Ensure when user A subscribed to the User-to-User service 1 implicit request calls user B subscribed to User-to-User service 1 a **User-to-user Information** parameter is present in the encapsulated ACM of the 180 response. |
| **Configuration** | User A is subscribed to the User-to-User service 1 implicit request |
| **SIP Parameter** | INVITE:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**IAM**<br>User-to-user Information<br>User Information<br><br>180/200<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**ACM/ANM**<br>User-to-user Information<br>User Information |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE (IAM) →<br>CASE A<br>← 180 Ringing (ACM)<br>CASE B<br>← 180 Ringing (ACM)<br>← 180 OK (ANM)<br>**Apply post test routine** | |
| **Comments** | **Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request?<br>**Check:** Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM?<br>**Check:** Is an ISUP/BICC ACM encapsulated in the 180 response?<br>**Check:** Is a User-to-user Information parameter present in the encapsulated ISUP/BICC ACM or ANM?<br>Repeat this test in reverse direction. |

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

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

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

| | |
|---|---|
| **Test case number** | SS_uus_006 |
| **Test case group** | SIP-SIP/SIP-I/UUS |
| **Reference** | 6.11.2, 7.1/ [24] |
| **SELECTION EXPRESSION** | ([Network A] SE 17 AND [Network A] SE 47) AND ([Network B] SE 17 AND [Network A] SE 47) AND SE 63 |
| **Test purpose** | **SIP-I support: Indicating of User-to-User service 1 essential explicit rejection.**<br><br>BICC/ISUP - SIP-I interworking applies in the originating and terminating network<br>User A is located in network A and user B is located in network B.<br>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:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**IAM**<br>User-to-user Indicator<br>Request<br>service 1<br>essential<br><br>500<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**REL**<br>Cause value<br>#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?<br>**Check:** Is a User-to-user Indicator parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 1', 'essential'?<br>**Check:** Is an ISUP/BICC REL encapsulated in the 500 response?<br>**Check:** Is the Cause value set to #29 or #69 in the encapsulated REL?<br>Repeat this test in reverse direction. |

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

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

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

| | |
|---|---|
| **Test case number** | SS_uus_010 |
| **Test case group** | SIP-SIP/SIP-I/UUS |
| **Reference** | 6.11.2, 7.1/ [24] |
| **SELECTION EXPRESSION** | ([Network A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND Network B] SE 47) AND SE 63 |
| **Test purpose** | **SIP-I support: Indicating of User-to-User service 2 essential rejection.**<br><br>BICC/ISUP - SIP-I interworking applies in the originating and terminating network<br>User A is located in network A and user B is located in network B.<br>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:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**IAM**<br>User-to-user Indicator<br>Request<br>service 2<br>essential<br><br>500<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**REL**<br>Cause value<br>#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?<br>**Check:** Is a User-to-user Indicator parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 1', 'essential'?<br>**Check:** Is an ISUP/BICC REL encapsulated in the 500 response?<br>**Check:** Is the Cause value set to #29 or #69 in the encapsulated REL?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_uus_011 |
| **Test case group** | SIP-SIP/SIP-I/UUS |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND Network A] SE 47 AND SE 63 |
| **Test purpose** | **SIP-I support: Indicating of User-to-User service 3 in initial INVITE request.**<br><br>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.<br>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:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**IAM**<br>User-to-user Indicator<br>Request<br>service 3<br>not essential or 'essential' |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE (IAM) ➔<br>**Apply post test routine** | |
| **Comments** | **Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request containing a User-to-user Indicator parameter, and the indicator **Request service 3** is set to the value 'not essential' or 'essential'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_uus_012 |
| **Test case group** | SIP-SIP/SIP-I/UUS |
| **Reference** | 5.4.3.2, 6.5, 7.1/ [24] |
| **SELECTION EXPRESSION** | ([Network A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND Network B] SE 47) AND SE 63 |
| **Test purpose** | **SIP-I support: Indicating of User-to-User service 3 in initial INVITE request successful.**<br><br>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.<br>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<br>User B is subscribed to the User-to-User service 3 |
| **SIP Parameter** | INVITE:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**IAM**<br>User-to-user Indicator<br>Request<br>service 3<br>not essential or 'essential'<br><br>200 OK<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**ANM**<br>User-to-user Indicator<br>Response<br>service 3 provided<br><br>INFO<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**USR**<br>User-to-user Information<br>User Information |
| **Message flow** | |

| **SIP (Network A)** | | **Interconnection Interface** | | **SIP (Network B)** |
|---|---|---|---|---|
| | | INVITE (IAM) | → | |
| | ← | 180 Ringing (ACM) | | |
| | ← | 200 OK INVITE (ANM) | | |
| | | ACK | → | |
| | | INFO (USR) | → | |
| | ← | 200 OK INFO | | |
| | ← | INFO (USR) | | |
| | | 200 OK INFO | → | |
| | | **Apply post test routine** | | |

| | |
|---|---|
| **Comments** | **Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request containing a User-to-user Indicator parameter, and the indicator **Request service 3** is set to the value 'not essential' or 'essential'?<br>**Check:** Is an ISUP/BICC ANM encapsulated in the 200 OK INVITE and the User-to-user Indicator parameter is set to 'Response', 'service 3 provided'?<br>**Check:** Is an ISUP/BICC USR encapsulated in the INFO message sent from network A to network B containing a User-to-user Information parameter?<br>**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?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_uus_013 |
| **Test case group** | SIP-SIP/SIP-I/UUS |
| **Reference** | 7.1, 6.5/ [24] |
| **SELECTION EXPRESSION** | ([Network A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND Network B] SE 47) AND SE 63 |
| **Test purpose** | **SIP-I support: Indicating of User-to-User service 3 not essential rejected in 200 OK response.**<br><br>BICC/ISUP - SIP-I interworking applies in the originating and terminating network<br>User A is located in network A and user B is located in network B.<br>Ensure when user A subscribed to the User-to-User service 3 not essential calls user B not subscribed to User-to-User service 3 the call is rejected by the network a **User-to-user Indicator** parameter is present set to 'Response', 'service 3 not provided' in the encapsulated ANM of the 200 OK final response. |
| **Configuration** | User A is subscribed to the User-to-User service 3<br>User B is not subscribed to the User-to-User service 3 |
| **SIP Parameter** | INVITE:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**IAM**<br>User-to-user Indicator<br>Request<br>service 3<br>not essential<br><br>200 OK<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**ANM**<br>User-to-user Indicator<br>Response<br>service 3 not provided |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE (IAM) →<br>← 180 Ringing (ACM)<br>← 200 OK INVITE (ANM)<br>ACK →<br>**Apply post test routine** |
| **Comments** | **Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request?<br>**Check:** Is a User-to-user Information parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 3' 'not essential'?<br>**Check:** Is an ISUP/BICC ANM encapsulated in the 200 OK response?<br>**Check:** Is a User-to-user Indicator parameter present set to 'Response', 'service 3 not provided' in the encapsulated ISUP/BICC ANM?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_uus_014 |
| **Test case group** | SIP-SIP/SIP-I/UUS |
| **Reference** | 6.11.2, 7.1/ [24] |
| **SELECTION EXPRESSION** | ([Network A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND Network B] SE 47) AND SE 63 |
| **Test purpose** | **SIP-I support: Indicating of User-to-User service 3 essential rejection.**<br><br>BICC/ISUP - SIP-I interworking applies in the originating and terminating network<br>User A is located in network A and user B is located in network B.<br>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:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**IAM**<br>User-to-user Indicator<br>Request<br>service 3<br>essential<br><br>500<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**REL**<br>Cause value<br>#29 or #69 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE (IAM) →<br>← 500 Server Internal Error (REL)<br>ACK →<br>**Apply post test routine** |
| **Comments** | **Check:** Is an ISUP/BICC IAM encapsulated in the initial INVITE request?<br>**Check:** Is a User-to-user Indicator parameter present in the encapsulated ISUP/BICC IAM set to 'Request', 'service 1', 'essential'?<br>**Check:** Is an ISUP/BICC REL encapsulated in the 500 response?<br>**Check:** Is the Cause value set to #29 or #69 in the encapsulated REL?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_uus_015 |
| **Test case group** | SIP-SIP/SIP-I/UUS |
| **Reference** | 5.4.3.2, 6.5, 7.1/ [24] |
| **SELECTION EXPRESSION** | ([Network A] SE 17 AND Network A] SE 47) AND ([Network B] SE 17 AND Network B] SE 47) AND SE 63 |
| **Test purpose** | **SIP-I support: Indicating of User-to-User service 3 during a session is established successful.**<br><br>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.<br>Ensure when user A subscribed to the User-to-User service 3 user A is able to request the User-to-User service 3 while the session is established. The User-to-User service is successful. |
| **Configuration** | User A is subscribed to the User-to-User service 3<br>User B is subscribed to the User-to-User service 3 |
| **SIP Parameter** | INFO:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**FAR**<br>Facility indicator<br>user-to-user service<br>User-to-user Indicator<br>Request<br>service 3<br>not essential<br><br>INFO:<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**FAA**<br>Facility indicator<br>user-to-user service<br>User-to-user Indicator<br>Response<br>service 3 provided<br><br>INFO<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**USR**<br>User-to-user Information<br>User Information |

**Message flow**

| **SIP (Network A)** | **Interconnection Interface** | | **SIP (Network B)** |
|---|---|---|---|
| | **A session is already established** | | |
| | INFO (FAR) | → | |
| ← | 200 OK INFO | | |
| ← | INFO (FAA) | | |
| | 200 OK INFO | → | |
| | | | |
| | INFO (USR) | → | |
| ← | 200 OK INFO | | |
| ← | INFO (USR) | | |
| | 200 OK INFO | → | |
| | **Apply post test routine** | | |

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

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

### 7.1.6.2 Subaddressing (SUB)

| | |
|---|---|
| **Test case number** | SS_sub_001 |
| **Test case group** | SIP-SIP/SIP-I/SUB |
| **Reference** | 7.1/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 62 |
| **Test purpose** | **SIP-I support: Calling party subaddress can be correctly transferred in the Access Transport parameters.**<br><br>BICC/ISUP - SIP-I interworking applies in the originating network User A is located in network A and user B is located in network B. Ensure that an ISUP/BICC ATP parameter present in the encapsulated IAM of the INVITE request and contains a Calling party subaddress. |
| **Configuration** | User A is subscribed to the SUB supplementary service |
| **SIP Parameter** | INVITE<br>Content-Type: multipart/mixed;boundary=[any boundary name]<br><br>--[any boundary name]<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**IAM**<br>Access transport<br>Calling party subaddress<br>--[any boundary name]-- |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE(**IAM**) →<br>**Apply post test routine** | |
| **Comments** | Establish a call from User A subscribed to the SUB supplementary service to user B<br>**Check:** Is an ISUP/BICC IAM present in the initial INVITE request?<br>**Check:** Is an ISUP/BICC ATP parameter present in the encapsulated IAM containing a Calling party subaddress?<br>Repeat this test in reverse direction. |

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

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

### 7.1.6.3 Terminal Portability (TP)

| | |
|---|---|
| **Test case number** | SS_tp_001 |
| **Test case group** | SIP-SIP/SIP-I/TP |
| **Reference** | 5.4.3.2/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 64 |
| **Test purpose** | **SIP-I support. SUS and RES messages transferred in an INFO request.**<br><br>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<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**SUS**<br>Suspend/resume indicator<br>ISDN subscriber initiated<br><br>INFO<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**RES**<br>Suspend/resume indicator<br>ISDN subscriber initiated |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>**A confirmed session already exists**<br>INFO(**SUS**) →<br>← 200 OK INFO<br><br>INFO(**RES**) →<br>← 200 OK INFO<br>**Apply post test routine** |
| **Comments** | A session is already established<br>**Check:** Is an ISUP **SUS** message encapsulated in the INFO request and the Suspend/resume indicator set to 'ISDN subscriber initiated'?<br>**Check:** Is an ISUP **RES** message encapsulated in the INFO request and the Suspend/resume indicator set to 'ISDN subscriber initiated'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_tp_002 |
| **Test case group** | SIP-SIP/SIP-I/TP |
| **Reference** | 5.4.3.2, 6.11.2, 6.11.2/ [24] |
| **SELECTION EXPRESSION** | [Network A] SE 17 AND [Network A] SE 47 AND SE 64 |
| **Test purpose** | **SIP-I support. SUS message transferred in an INFO request call released.**<br><br>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<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**SUS**<br>Suspend/resume indicator<br>ISDN subscriber initiated<br><br>BYE<br>Content-Type: application/isup;version=itu-t92<br>Content-Disposition: signal;handling=required<br>**REL**<br>Location<br>public network serving remote user<br>Cause value<br>102 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>**A confirmed session already exists**<br>INFO(**SUS**) →<br>← 200 OK INFO<br><br>BYE(**REL**) →<br>← 200 OK BYE |
| **Comments** | A session is already established<br>**Check:** Is an ISUP **SUS** message encapsulated in the INFO request and the Suspend/resume indicator set to ISDN 'subscriber initiated'?<br>**Check:** Is an ISUP **REL** message encapsulated in the BYE request and the Cause value set to #102?<br>Repeat this test in reverse direction. |

## 7.2 Number Portability

| | |
|---|---|
| **Test case number** | SS_NP_001 |
| **Test case group** | SIP-SIP/NubP |
| **Reference** | 5.3, 5.4/ [2] |
| **SELECTION EXPRESSION** | [Network A] SE 13 |
| **Test purpose** | **Request line in the INVITE contains the number portability indication.**<br><br>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<br>sip: + <CC> <NDC> <SN>[;npdi][; rn=(Number portability routing number)]<br>@<hostname>;user = phone SIP/2.0 |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE ➔<br>**Apply post test routine** | |
| **Comments** | **Check:** Is the URI in the userinfo of the Request line in a global number format?<br>**Check:** Is the URI rn parameter containing the Number Portability Routing Number in a global number format?<br>**Check:** Is optional the URI parameter 'npdi' present?<br>**Check:** Is the user parameter set to 'phone'?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_NP_002 |
| **Test case group** | SIP-SIP/NubP |
| **Reference** | 5.3, 5.4/ [2] |
| **SELECTION EXPRESSION** | NOT [Network A] SE 13 |
| **Test purpose** | **Request line in the INVITE without npdi parameter.**<br><br>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<br>sip: + <CC> <NDC> <SN>@<hostname>;user = phone SIP/2.0 |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE ➔<br>**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?<br>**Check:** Is the user parameter set to 'phone'?<br>Repeat this test in reverse direction. |

## 7.3 Accounting

| | |
|---|---|
| **Test case number** | SS_acc_001 |
| **Test case group** | SIP-SIP/ACCOUNTING |
| **Reference** | |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Comparison of Charging Data Records > 1 s.**<br><br>Accounting of a confirmed session with a duration > 1 s. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>← 200 OK INVITE<br>ACK →<br>**Communication**<br>BYE →<br>← 200 OK BYE |
| **Comments** | 1. Setup a call from network A to network B.<br>2. Verify is the session confirmed.<br>3. Terminate the session after 5 s.<br>4. Determine the duration of the session from the trace of the call monitor.<br>5. Compare the following information elements indicated in the CDR's of both networks:<br>• calling party number<br>• called party number<br>• timestamp<br>• callduration<br>• callsetuptime (optional)<br>6. Check the duration indicated in the CDR against the duration in the call trace.<br>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**<br><br>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**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>← 200 OK INVITE<br>ACK →<br>**Communication**<br>BYE →<br>← 200 OK BYE | |
| **Comments** | 1. Setup a call from network A to network B.<br>2. Verify is the session confirmed.<br>3. Terminate the session after 5 s.<br>4. Determine the duration of the session from the trace of the call monitor.<br>5. Compare the following information elements indicated in the CDR's of both networks:<br>• calling party number<br>• called party number<br>• timestamp<br>• callduration<br>• callsetuptime (optional)<br>6. Check the duration indicated in the CDR against the duration in the call trace.<br>7. Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_acc_003 |
| **Test case group** | SIP-SIP/ACCOUNTING |
| **Reference** | |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Comparison of Charging Data Records > 15 min.**<br><br>Accounting of a confirmed session with a duration of > 15 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>← 200 OK INVITE<br>ACK →<br>**Communication**<br>BYE →<br>← 200 OK BYE |
| **Comments** | 1. Setup a call from network A to network B.<br>2. Verify is the session confirmed.<br>3. Terminate the session after 15 min.<br>4. Determine the duration of the session from the trace of the call monitor.<br>5. Compare the following information elements indicated in the CDR's of both networks:<br>• calling party number<br>• called party number<br>• timestamp<br>• callduration<br>• callsetuptime (optional)<br>6. Check the duration indicated in the CDR against the duration in the call trace.<br>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 min.**<br><br>Accounting of a confirmed session with a duration of 25 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>← 200 OK INVITE<br>ACK →<br>**Communication**<br>BYE →<br>← 200 OK BYE | |
| **Comments** | 1. Setup a call from network A to network B.<br>2. Verify is the session confirmed.<br>3. Terminate the session after 25 min.<br>4. Determine the duration of the session from the trace of the call monitor.<br>5. Compare the following information elements indicated in the CDR's of both networks:<br>• calling party number<br>• called party number<br>• timestamp<br>• callduration<br>• callsetuptime (optional)<br>6. Check the duration indicated in the CDR against the duration in the call trace.<br>7. Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_acc_005 |
| **Test case group** | SIP-SIP/ACCOUNTING |
| **Reference** | |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Comparison of Charging Data Records more than 30 min.**<br><br>Accounting of a confirmed session with a duration of > 30 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow** | |

| SIP (Network A) | | Interconnection Interface | | SIP (Network B) |
|---|---|---|---|---|
| | | INVITE | → | |
| | ← | 180 Ringing | | |
| | ← | 200 OK INVITE | | |
| | | ACK | → | |
| | | **Communication** | | |
| | | BYE | → | |
| | ← | 200 OK BYE | | |

| | |
|---|---|
| **Comments** | 1. Setup a call from network A to network B.<br>2. Verify is the session confirmed.<br>3. Terminate the session after 35 min.<br>4. Determine the duration of the session from the trace of the call monitor.<br>5. Compare the following information elements indicated in the CDR's of both networks:<br>• calling party number<br>• called party number<br>• timestamp<br>• callduration<br>• callsetuptime (optional)<br>6. Check the duration indicated in the CDR against the duration in the call trace.<br>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 min.**<br><br>Accounting of a confirmed session with a duration between 60 min and 120 min. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>← 200 OK INVITE<br>ACK →<br>**Communication**<br>BYE →<br>← 200 OK BYE |
| **Comments** | 1. Setup a call from network A to network B.<br>2. Verify is the session confirmed.<br>3. Terminate the session at the earliest 61 min and at the latest 119 min.<br>4. Determine the duration of the session from the trace of the call monitor.<br>5. Compare the following information elements indicated in the CDR's of both networks:<br>• calling party number<br>• called party number<br>• timestamp<br>• callduration<br>• callsetuptime (optional)<br>6. Check the duration indicated in the CDR against the duration in the call trace.<br>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.**<br><br>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**<br>**SIP (Network A)** | **Interconnection Interface**<br>INVITE →<br>← 180 Ringing<br>← 200 OK INVITE<br>ACK →<br>**Communication**<br>BYE →<br>← 200 OK BYE<br>**SIP (Network B)** |
| **Comments** | 1. Setup a call from network A to network B.<br>2. Verify is the session confirmed.<br>3. Terminate the session after 24 hours.<br>4. Determine the duration of the session from the trace of the call monitor.<br>5. Compare the following information elements indicated in the CDR's of both networks:<br>• calling party number<br>• called party number<br>• timestamp<br>• callduration<br>• callsetuptime (optional)<br>6. Check the duration indicated in the CDR against the duration in the call trace.<br>7. Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_acc_007A |
| **Test case group** | SIP-SIP/ACCOUNTING |
| **Reference** | |
| **SELECTION EXPRESSION** | |
| **Test purpose** | **Comparison of Charging Data Records less than 1 s.**<br><br>Accounting of a confirmed session with duration <1 s. Verify the duration of the active session stored in the CDR of both networks compared with the duration in the monitored message flow at the Interconnection Interface. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br>← 200 OK INVITE<br>ACK →<br>**Communication**<br>BYE →<br>← 200 OK BYE |
| **Comments** | 1. Set up a call from Network A to Network B.<br>2. Verify whether the session confirmed.<br>3. Terminate the session after 0,9 s.<br>4. Determine the duration of the session from the trace of the call monitor.<br>5. Compare the following information elements indicated in the CDRs of both networks:<br>• calling party number<br>• called party number<br>• timestamp<br>• call duration<br>• call setup time (optional).<br>6. Check the duration indicated in the CDR against the duration in the call trace.<br>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 session not confirmed.**<br><br>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**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 180 Ringing<br><br>BYE/CANCEL →<br>← 200 OK BYE/CANCEL<br>← 487 Request Terminated<br>ACK → | |
| **Comments** | 1. Setup a call from network A to network B.<br>2. Verify is an early dialogue established.<br>3. Terminate the early dialogue after 20 s.<br>4. Determine the duration of the session from the trace of the call monitor.<br>5. Compare the following information elements indicated in the CDR's of both networks:<br>• calling party number<br>• called party number<br>• timestamp<br>• callduration<br>• callsetuptime (optional)<br>6. Check the duration indicated in the CDR against the duration in the call trace.<br>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/ [2] |
| **SELECTION EXPRESSION** | [Network A] SE14 AND [Network B] SE15 |
| **Test purpose** | **User selects an operator 'call-by-call'.**<br><br>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<br>sip: + <CC> <NDC> <SN>**[**;cic=(carrier ID)**]**@<hostname> user=phone SIP/2.0<br><br>INVITE: Request line<br>sip: + <CC> <NDC> <SN>;npdi<br>[;rn=<Number portability routing number>]@<hostname>;<br>user=phone SIP/2.0 |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE 1 →<br>← INVITE 2<br>**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?<br>**Check:** Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A?<br>**Check:** Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A?<br>**NOTE 1:** The 'cic' parameter may be absent according national regulations or national agreements.<br>**NOTE 2:** It is possible that further information is available in the Request line regarding the end user charging in case of Carrier selection.<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_csel_002 |
| **Test case group** | SIP-SIP/CS |
| **Reference** | 5.7.1.10/ [2] |
| **SELECTION EXPRESSION** | [Network A] SE14 AND [Network B] SE15 |
| **Test purpose** | **User is presubscribed to operator B.**<br><br>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<br>sip: + <CC> <NDC> <SN>**[**;cic=(carrier ID)**]**@<hostname> user=phone SIP/2.0<br><br>INVITE: Request line<br>sip: + <CC> <NDC> <SN>;npdi<br>[;rn=<Number portability routing number>]@<hostname>;<br>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?<br>**Check:** Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A?<br>**Check:** Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A?<br>**NOTE 1:** The 'cic' parameter may be absent according national regulations or national agreements.<br>**NOTE 2:** It is possible that further information is available in the Request line regarding the end user charging in case of Carrier selection.<br>Repeat this test in reverse direction. |

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

| | |
|---|---|
| **Test case number** | SS_csel_004 |
| **Test case group** | SIP-SIP/CS |
| **Reference** | 5.7.1.10/ [2] |
| **SELECTION EXPRESSION** | [Network A] SE14 AND [Network B] SE15 |
| **Test purpose** | **User is presubscribed to operator B, and overrides the preselection with call-by-call via operator B.**<br><br>User A and user B are located in network A. User A is preselected to network B. Ensure that user A is able to call user B and user A is able to select network B as a selected carrier 'call-by-call'. The preselected carrier is ignored. |
| **Configuration** | User in network A is presubscribed to network B |
| **SIP Parameter** | INVITE: Request line<br>sip: + <CC> <NDC> <SN>**[**;cic=(carrier ID)**]**@<hostname> user=phone SIP/2.0<br><br>INVITE: Request line<br>sip: + <CC> <NDC> <SN>;npdi<br>[;rn=<Number portability routing number>]@<hostname>;<br>user=phone SIP/2.0 |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE 1 →<br>← INVITE 2<br>**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?<br>**Check:** Is the 'npdi' parameter present in the Request URI of the INVITE request sent from network B to network A?<br>**Check:** Is optional the 'rn' parameter present in the Request URI of the INVITE request sent from network B to network A?<br>**NOTE 1:** The 'cic' parameter may be absent according national regulations or national agreements.<br>**NOTE 2:** It is possible that further information is available in the Request line regarding the end user charging in case of Carrier selection.<br>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.**<br><br>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<br>Users in network A are in the same CUG |
| **SIP Parameter** | INVITE: Request line<br>sip: + <CC> <NDC> <SN>@ <hostname><br>user=phone SIP/2.0<br><br>Content-Type: application/vnd.etsi.cug+xml<br>Content-Disposition: ….;handling= required<br><br>…….<br><…:cug><br>……<br><..: cugCommunicationIndicator>11</…: cugCommunicationIndicator><br><…:cug><br><br>INVITE: Request line<br>sip: + <CC> <NDC> <SN@<hostname>;user=phone SIP/2.0<br><br>Content-Type: application/vnd.etsi.cug+xml<br>Content-Disposition: ….;handling= required<br><br>…….<br><…:cug><br>……<br><..: cugCommunicationIndicator>11</…: cugCommunicationIndicator><br><…:cug> |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE 1 →<br>← INVITE 2<br>**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?<br>**Check:** Is optional the 'rn' parameter present in the userinfo of the INVITE request sent from network B to network A?<br>**Check:** Contains the XML body in the INVITE a 'cugCommunicationIndicator' element set to '11' as a 'cug' child element?<br>**Check:** Is the session setup not rejected? |

## 7.5 Emergency call

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

# 7.6 SIP Support of Charging

| | |
|---|---|
| **Test case number** | SS_sipc_001 |
| **Test case group** | SIP-SIP/ SIP_charging |
| **Reference** | B.2.3/ [19] |
| **SELECTION EXPRESSION** | SE 16 |
| **Test purpose** | **Successful session from user A to user B via network B one single tariff.**<br><br>User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a tariff information with one single tariff covered in a XML MIME body in a reliable provisional or successful final response. |
| **Configuration** | |
| **SIP Parameter** | INVITE:<br>Supported: 100rel<br><br>18x or 200 OK<br>Require: 100rel<br>ContentType: application/vnd.etsi.sci+xml<br>Content-Disposition: render; handling=optional<br><br>messageType<br>crgt<br>chargingControlIndicators<br>chargingTariff<br>tariffCurrency<br>currentTariffCurrency<br>communicationChargeSequenceCurrency<br>currencyFactorScale<br>currencyFactor<br>currencyScale<br>tariffDuration<br>subTariffControl<br>tariffControlIndicators<br>originationIdentification<br>currency (optional) |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>**CASE A** ← 18x(crgt)<br>PRACK →<br>← 200 OK PRACK<br><br>**CASE B** ← 200 OK INVITE(crgt)<br>**Apply post test routine** | |
| **Comments** | **Check:** Is the supported header in the initial INVITE set to '100rel'<br>**Check:** Is the Require header in the response containing the tariff information set to '100rel'?<br>**Check:** Is the messageType '**crgt**' present in a 1xx provisional or a 200 OK INVITE final response?<br>**Check:** Is the **tariffCurrency** element set to 'currentTariffCurrency'?<br>**Check:** Represents the **currencyFactorScale** in the communicationChargeSequenceCurrency element the applicable tariff?<br>**Check:** Is the tariffDuration element set to '0'?<br>**Check:** Is the optional element '**currency**' set to 'EUR' if present?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_sipc_002 |
| **Test case group** | SIP-SIP/ SIP_charging |
| **Reference** | B.2.3/ [19] |
| **SELECTION EXPRESSION** | SE 16 |
| **Test purpose** | **Successful session from user A to user B via network B several tariffs in one sequence.**<br><br>User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a tariff information with several tariffs in a sequence covered in a XML MIME body in a reliable provisional or successful final response. |
| **Configuration** | |
| **SIP Parameter** | INVITE:<br>Supported: 100rel<br><br>18x or 200 OK<br>Require: 100rel<br>ContentType: application/vnd.etsi.sci+xml<br>Content-Disposition: render; handling=optional<br><br>messageType<br>crgt<br>chargingControlIndicators<br>chargingTariff<br>tariffCurrency<br>currentTariffCurrency<br>communicationChargeSequenceCurrency<br>currencyFactorScale<br>currencyFactor<br>currencyScale<br>tariffDuration<br>subTariffControl<br>communicationChargeSequenceCurrency<br>currencyFactorScale<br>currencyFactor<br>currencyScale<br>tariffDuration<br>subTariffControl<br>tariffControlIndicators<br>originationIdentification<br>currency (optional) |
| **Message flow**<br>**SIP (Network A)**<br><br>**CASE A**<br><br><br><br>**CASE B** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 18x(crgt)<br>PRACK →<br>← 200 OK PRACK<br><br>← 200 OK INVITE(crgt)<br>**Apply post test routine** |
| **Comments** | **Check:** Is the Supported header in the initial INVITE set to '100rel'?<br>**Check:** Is the Require header in the response containing the tariff information set to '100rel'?<br>**Check:** Is the messageType '**crgt**' present in a 1xx provisional or a 200 OK INVITE final response?<br>**Check:** Is the **tariffCurrency** element set to 'currentTariffCurrency'?<br>**Check:** Are there **more than one communicationCharge SequenceCurrency** elements present in the currentTariffCurrency element?<br>**Check:** Represents the **currencyFactorScale** in the communicationCharge SequenceCurrency elements the applicable tariffs?<br>**Check:** Is the tariffDuration element in the last applicable tariff set to '0'?<br>**Check:** Is the optional element '**currency**' set to 'EUR' if present?<br>Repeat this test in reverse direction. |

| | |
|---|---|
| **Test case number** | SS_sipc_003 |
| **Test case group** | SIP-SIP/ SIP_charging |
| **Reference** | B.2.3/ [19] |
| **SELECTION EXPRESSION** | SE 16 |
| **Test purpose** | **Successful session from user A to user B via network B with call attempt charge.**<br><br>User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a tariff information with a call attempt charge covered in a XML MIME body in a reliable provisional or successful final response. |
| **Configuration** | |
| **SIP Parameter** | INVITE:<br>Supported: 100rel<br><br>18x or 200 OK<br>Require: 100rel<br>ContentType: application/vnd.etsi.sci+xml<br>Content-Disposition: render; handling=optional<br><br>messageType<br>crgt<br>chargingControlIndicators<br>chargingTariff<br>tariffCurrency<br>currentTariffCurrency<br>communicationChargeSequenceCurrency<br>currencyFactorScale<br>currencyFactor<br>currencyScale<br>tariffDuration<br>subTariffControl<br>tariffControlIndicators<br>callAttemptChargeCurrency<br>currencyFactor<br>currencyScale<br>originationIdentification<br>currency (optional) |
| **Message flow**<br>**SIP (Network A)** **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>**CASE A** ← 18x(crgt)<br>PRACK →<br>← 200 OK PRACK<br><br>**CASE B** ← 200 OK INVITE(crgt)<br>**Apply post test routine** | |
| **Comments** | **Check:** Is the supported header in the initial INVITE set to '100rel'?<br>**Check:** Is the Require header in the response containing the tariff information set to '100rel'?<br>**Check:** Is the messageType a '**crgt**' present in a 1xx provisional or a 200 OK INVITE final response?<br>**Check:** Is the **tariffCurrency** element set to 'callAttemptChargeCurrency'?<br>**Check:** Represents the **currencyFactorScale** in the **callAttemptChargeCurrency** element the applicable tariff?<br>**Check:** Is the optional element '**currency**' set to 'EUR' if present?<br>Repeat this test in reverse direction. |

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

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

| | |
|---|---|
| **Test case number** | SS_sipc_006 |
| **Test case group** | SIP-SIP/ SIP_charging |
| **Reference** | B.2.3/ [19] |
| **SELECTION EXPRESSION** | SE 16 |
| **Test purpose** | **Successful change of a current tariff and next tariff during an active session.**<br><br>User A is located in network A and network B is responsible for charging (CDP) in case of carrier selection or service. Ensure that the network B sends a new tariff information with several current tariffs and several next tariffs covered in a XML MIME body in an INFO request. |
| **Configuration** | |
| **SIP Parameter** | INFO<br>    ContentType: application/vnd.etsi.sci+xml<br><br>    messageType<br>    crgt<br>        chargingControlIndicators<br>        chargingTariff<br>            tariffCurrency<br>                currentTariffCurrency<br>                    communicationChargeSequenceCurrency<br>                        currencyFactorScale<br>                            currencyFactor<br>                            currencyScale<br>                        tariffDuration<br>                        subTariffControl<br>                    communicationChargeSequenceCurrency<br>                        currencyFactorScale<br>                            currencyFactor<br>                            currencyScale<br>                        tariffDuration<br>                        subTariffControl<br>                    tariffControlIndicators<br>                tariffSwitchCurrency<br>                    nextTariffCurrency<br>                        communicationChargeSequenceCurrency<br>                            currencyFactorScale<br>                                currencyFactor<br>                                currencyScale<br>                            tariffDuration<br>                            subTariffControl<br>                        communicationChargeSequenceCurrency<br>                            currencyFactorScale<br>                                currencyFactor<br>                                currencyScale<br>                            tariffDuration<br>                            subTariffControl<br>                        tariffControlIndicators<br>                    tariffSwitchOverTime<br>        originationIdentification<br>        currency (optional) |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>**A confirmed session already exists**<br>← INFO<br>200 OK INFO →<br>**Apply post test routine** |
| **Comments** | **Check:** Is the messageType '**crgt**' present in the INFO request?<br>**Check:** Is the **tariffCurrency** element set to 'currentTariffCurrency'?<br>**Check:** Represents the **currencyFactorScale** in the communicationChargeSequenceCurrency elements the current tariffs?<br>**Check:** Is the **tariffSwitchCurrency** element set to 'nextTariffCurrency'?<br>**Check:** Represents the **currencyFactorScale** in the communicationChargeSequenceCurrency elements the next tariffs?<br>Repeat this test in reverse direction. |

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

# 7.7 Quality of Service (optional)

## 7.7.1 Delay Values

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

## 7.7.2 Test purposes for Quality of Service test

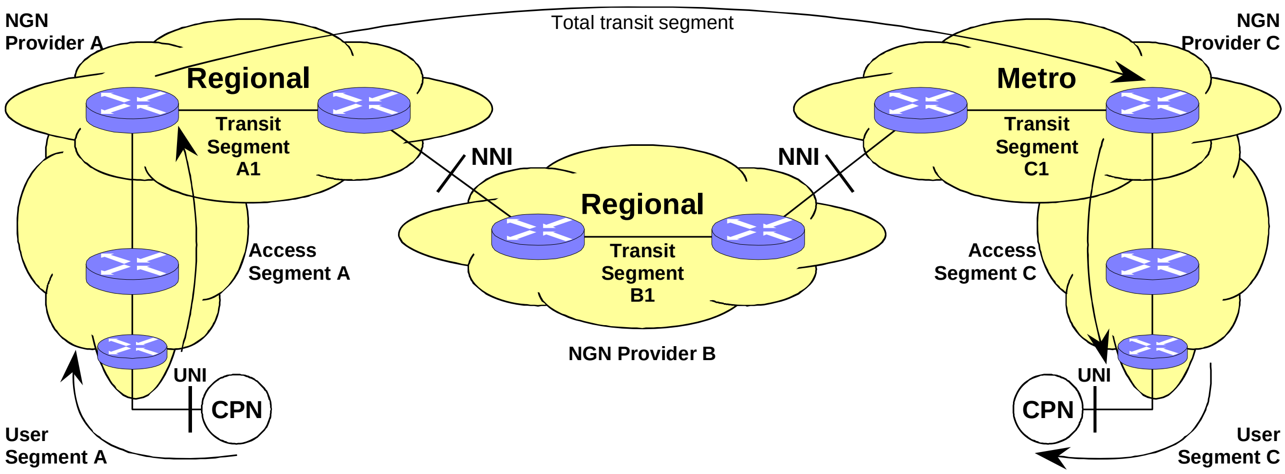


**Figure 7.7.2-1 : General Reference Configuration**

| | |
|---|---|
| **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.<br>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.<br>Based on the measurement determine the transit segment delay.<br>The call is released from the calling user. |
| **Configuration** | The amplitude of the tone is -16 dBm0;<br>Minimum uplink/downlink bandwidth is 1 Mbit/s |
| **SIP Parameter** | |
| **Message flow**<br>**SIP (Network A)** | **Interconnection Interface** **SIP (Network B)**<br>INVITE →<br>← 100 Trying<br>← 180 Ringing<br>← 200 OK INVITE<br>ACK →<br>Communication<br>→ **BYE**<br>200 OK BYE ← |
| **Comments** | • UE1 (a) establishes call to UE2 (b).<br>• Call answered and held for 80 seconds.<br>• Quality assessed |

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

| | |
|---|---|
| **Test case number** | SS_qos_003 |
| **Test case group** | SIP-SIP/QoS |
| **Reference** | [4] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | Ensure that the UE can successfully activate the voice call via dedicated voice bearer.<br>The test call is successful in the case if the Call setup time (PDD) does not exceed the values listed in table 7.2.2-1 and call is stable in unanswered and answered phases, the call remains in intelligible/high quality conversation phase for 80 seconds.<br>The call is released from the calling user. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow** | |

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

| | |
|---|---|
| **Comments** | • UE1 (a) establishes call to UE2 (b).<br>• Call answered and held for 80 seconds.<br>• Quality assessed<br>Repeat this test in reverse direction. |

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

| Meaning of timers | Parameter Recommendation ITU-T Q.543 [31] | IMS, PES equivalent | Reference Load A | | Reference Load B | |
|---|---|---|---|---|---|---|
| | Detailed description | | Mean Value | 95 % probability of not exceeding | Mean Value | 95 % probability of not exceeding |
| VoLTE -VoLTE [29]<br>IMS to VoLTE | | | | | | |
| Call setup time: The definition of Call setup time for VoLTE is defined in ETSI TS 102 250-2 [32]. | | | | | | |
| | | | ≤ 1 950 ms | ≤ 2 100 ms | ≤ 2 250 ms | ≤ 2 400 ms<br>Note 1<br>Note 2<br>Note 3 |
| VoLTE to IMS (note 4) | | | | | | |
| Call setup time (post dialling delay, PDD [30]<br>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. | | | | | | |
| | | | ≤ 420 ms | ≤ 580 ms | ≤ 750 ms | ≤ 900 ms<br>Note 4 |
| IMS - IMS | | | | | | |
| Call setup time (post dialling delay, PDD) | | | | | | |
| | | | ≤ 350 ms | ≤ 500 ms | ≤ 650 ms | ≤ 800 ms |
| NOTE 1: Paging Cycle 1,28 s.<br>NOTE 2: S1-Control plane delay: 2 ms - 15 ms (S1 is the interface between eNode Bs and MME and S-GW).<br>NOTE 3: The maximum value should not exceed 5,9 seconds [29].<br>NOTE 4: The values are based on the condition that the originating VoLTE - UE is in their state ECM Connected. In the case when the VoLTE - UE is in their state ECM Idle, the time duration is about 100 ms higher. | | | | | | |
| ISDN-ISDN | | | | | | |
| FFS | | | | | | |

| | |
|---|---|
| **Test case number** | SS_qos_004 |
| **Test case group** | SIP-SIP/QoS |
| **Reference** | [4] |
| **SELECTION EXPRESSION** | |
| **Test purpose** | Ensure that the UE can successfully activate the voice call via dedicated voice bearer.<br>The test call is successful if the call remains in intelligible/high quality conversation phase for 80 seconds.<br>The call is released from the called user. |
| **Configuration** | |
| **SIP Parameter** | |
| **Message flow** | |

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

| | |
|---|---|
| **Comments** | • UE1 (a) establishes call to UE2 (b).<br>• Call answered and held for 80 seconds.<br>• Quality assessed<br>Repeat this test in reverse direction. |

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

## A.0 Introduction

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

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

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

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

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

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

## A.2 Second step - Selection Expression

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

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

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

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

# A.3 Third step - Access and End Devises Types

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

**Table A.3: Overview of device types with examples**

List of Type of End devices in both networks

| | Network A<br>Telekom Austria | Network B<br>Deutsche Telekom |
|---|---|---|
| SIP-VoIP | | x |
| POTS | | x |
| ISDN | | |
| GSM | | |
| VoUMTS | | |
| VoLTE | | |
| PSTN | x | x |

| Highlight color | Explanation | Refernce |
|---|---|---|
| | The user equipment is a SIP hardphone or a SIP soft client on a PC in the fixed network<br>The user equipment is a 4G mobile device in an LTE network<br>The user equipment is a 3G mobile device in an UMTS network | TS 124 229 |
| | The user equipment is an integrated end device in the fixed network - access via a legacy analogue device | TS 183 043 |
| | The user equipment is an integrated end device in the fixed network - access via a legacy ISDN device | TS 183 036 |
| | The user equipment is a 2G mobile device in an GSM network. SS7 / SIP interworking applies<br>The user equipment is located in a fixed SS7 network (analogue or ISDN) | ITU-T Q.761 - Q764<br>TS129 163<br>ITU-T Q.1912.5 |

# A.4 Fourth step - activation

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

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

**Table A.4: Test list - example**

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

# Annex B (informative): Bibliography


- IETF RFC 3966 (2004): "The tel URI for Telephone Numbers".
- IETF RFC 3311 (2002): "The Session Initiation Protocol (SIP) UPDATE Method".
- IETF RFC 3323 (2002): "A Privacy Mechanism for the Session Initiation Protocol (SIP)".
- IETF RFC 3325 (2002): "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks".
- IETF RFC 2833: "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".
- ETSI TS 134 229-1: "Universal Mobile Telecommunications System (UMTS); Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Part 1: Protocol conformance specification (3GPP TS 34.229-1 version 6.3.0 Release 6)".
- ETSI EG 201 018: "Integrated Services Digital Network (ISDN); Application of the Bearer Capability (BC), High Layer Compatibility (HLC) and Low Layer Compatibility (LLC) information elements by terminals supporting ISDN services".
- ETSI EN 300 093-1: "Integrated Services Digital Network (ISDN); Calling Line Identification Restriction (CLIR) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 207-1: "Integrated Services Digital Network (ISDN); Diversion supplementary services; Digital Subscriber Signalling System No. One (DSS1); Part 1: Protocol specification".
- ETSI EN 300 188-1: "Integrated Services Digital Network (ISDN); Three-Party (3PTY) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 141-1: "Integrated Services Digital Network (ISDN); Call Hold (HOLD) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 185-1: "Integrated Services Digital Network (ISDN); Conference call, add-on (CONF) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 196-1: "Integrated Services Digital Network (ISDN); Generic functional protocol for the support of supplementary services; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 138-1: "Integrated Services Digital Network (ISDN); Closed User Group (CUG) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI TS 124 147: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; Stage 3 (3GPP TS 24.147 version 9.1.0 Release 9)".
- ETSI EN 300 001: "Attachments to the Public Switched Telephone Network (PSTN); General technical requirements for equipment connected to an analogue subscriber interface in the PSTN".
- ETSI ETS 300 648: "Public Switched Telephone Network (PSTN); Calling Line Identification Presentation (CLIP) supplementary service; Service description".
- ETSI EN 300 092-1: "Integrated Services Digital Network (ISDN); Calling Line Identification Presentation (CLIP) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol; Part 1: Protocol specification".
- ETSI EN 300 659: "Access and Terminals (AT); Analogue access to the Public Switched Telephone Network (PSTN); Subscriber line protocol over the local loop for display (and related) services".

- ETSI TBR 008: "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice; Attachment requirements for handset terminals".
- Recommendation ITU-T Q.951: "Stage 3 description for number identification supplementary services using DSS 1".
- Recommendation ITU-T Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".
- Recommendation ITU-T Q.850 (05/98): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".
- ETSI EG 201 299-1: "Integrated Services Digital Network (ISDN); Network Integration Testing (NIT); ISDN/PSTN end-to-end testing; Part 1: Test Suite Structure and Test Purposes (TSS&TP) specification".

# History

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