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Technical Specification

Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks;

Part 2: Test Suite Structure and Test Purposes (TSS&TP)

Reference
DTS/TISPAN-06025-2-NGN-R2

Keywords
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Contents

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN).

The present document is part 2 of a multi-part deliverable covering the Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks;

Part 1: "Protocol Implementation Conformance Statement (PICS)";

Part 2: "Test Suite Structure and Test Purposes (TSS&TP)";

Part 3: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) proforma specification".

1 Scope

The present document specifies the Test Suite Structure and Test Purposes (TSS&TP) for SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN)subsystem and Circuit Switched (CS) networks ES 283 027 [1]. The references [1] and [16] are indentical.

A further part of the present document specifies the Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) proforma based on the present document.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

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2.1 Normative references

The following referenced documents are indispensable for the application of the present document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

[1] ETSI ES 283 027 (V2.5.1): "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN) Endorsement of the SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks [3GPP TS 29.163 (Release 7), modified]".

[2] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+) Universal Mobile Telecommunications System (UMTS) Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 version 7.9.0 Release 7)".

[3] ITU-T Recommendations Q.761 to Q.764 (2000): "Signalling System No.7 ISDN User Part (ISUP)".

[4] Void.

[5] ITU-T Recommendation Q.850 (1998): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".

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

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

[8] ISO/IEC 9646-1 (1994): "Conformance testing methodology and framework - Part 1: General Concepts".

[9] ISO/IEC 9646-3 (1992): "Conformance testing methodology and framework - Part 3: The Tree and Tabular Combined Notation".

[10] ISO/IEC 9646-7 (1994): "Conformance testing methodology and framework - Part 7: Implementation Conformance Statement".

[11] ITU-T Recommendation E.164: "The international public telecommunication numbering plan".

[12] Void.

[13] ITU-T Recommendation Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".

[14] ETSI TS 183 008: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR); Protocol specification".

[15] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 version 8.5.0 Release 8)".

[16] ETSI TS 129 527 (V8.2.0): "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); TISPAN; Endorsement of the SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks [3GPP TS 29.163 (Release 7), modified] (3GPP TS 29.527 version 8.2.0 Release 8)".

[17] IETF RFC 3556: "Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) Bandwidth".

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

[19] IETF RFC 4040: "RTP Payload Format for a 64 kbit/s Transparent Call".

[20] ITU-T Recommendation F.182: "Operational provisions for the international public facsimile service between subscribers with Group 3 facsimile terminals (Telefax 3)".

[21] ITU-T Recommendation F.184: "Operational provisions for the international public facsimile service between subscriber stations with group 4 facsimile terminals (telefax 4)".

[22] ITU-T Recommendation F.230: "Service requirements unique to the mixed mode (MM) used within the teletex service".

[23] ITU-T Recommendation F.220: "Service requirements unique to the processable mode number eleven (PM11) used within the teletex service".

[24] ITU-T Recommendation F.200: "Teletex service".

[25] ITU-T Recommendation F.300: "Videotex service".

[26] ITU-T Recommendation F.60: "Operational provisions for the international telex service".

[27] ITU-T Recommendation F.721: "Videotelephony teleservice for ISDN".

[28] ETSI ETS 300 356-1: "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP) version 4 for the international interface; Part 1: Basic services [ITU-T Recommendations Q.761 to Q.764 (1999) modified]".

[29] ITU-T Recommendation X.213: "Information technology - Open Systems Interconnection - Network service definition".

[30] ISO/IEC 8348: "Information technology - Open Systems Interconnection - Network service definition".

[31] ITU-T Recommendation T.38: "Procedures for real-time Group 3 facsimile communication over IP networks".

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

[33] ITU-T Recommendation Q.737.1: "Stage 3 description for additional information transfer supplementary services using Signalling System No. 7 : User-to-user signalling (UUS)".

[34] ITU-T Recommendation Q.734.1: "Stage 3 description for multiparty supplementary services using Signalling System No. 7: Conference calling".

[35] ITU-T Recommendation Q.734.2: "Stage 3 description for multiparty supplementary services using Signalling System No. 7: Three-party service".

2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area**.** For non-specific references, the latest version of the referenced document (including any amendments) applies.

Not applicable.

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in SIP / ISUP interworking reference specification, in ISDN layer 3 reference specification, in ISO/IEC 9646-1 [8], in ISO/IEC 9646-3 [9], in ISO/IEC 9646-7 [10] and the following apply:

Abstract Test Case (ATC): complete and independent specification of the actions required to achieve a specific test purpose, defined at the level of abstraction of a particular Abstract Test Method, starting in a stable testing state and ending in a stable testing state

Abstract Test Method (ATM): description of how an SUT is to be tested, given at an appropriate level of abstraction to make the description independent of any particular realization of a Means of Testing, but with enough detail to enable abstract test cases to be specified for this method

Abstract Test Suite (ATS): test suite composed of abstract test cases

Implementation Under Test (IUT): implementation of one or more OSI protocols in an adjacent user/provider relationship, being part of a real open system which is to be studied by testing

Means of Testing (MOT): combination of equipment and procedures that can perform the derivation, selection, parameterization and execution of test cases, in conformance with a reference standardized ATS, and can produce a conformance log

PICS proforma: document, in the form of a questionnaire, which when completed for an implementation or system becomes the PICS

PIXIT proforma: document, in the form of a questionnaire, which when completed for the SUT becomes the PIXIT

Point of Control and Observation (PCO): point within a testing environment where the occurrence of test events is to be controlled and observed, as defined in an Abstract Test Method

pre-test condition: setting or state in the SUT which cannot be achieved by providing stimulus from the test environment

Protocol Implementation Conformance Statement (PICS): statement made by the supplier of a protocol claimed to conform to a given specification, stating which capabilities have been implemented

Protocol Implementation eXtra Information for Testing (PIXIT): statement made by a supplier or implementor of an SUT (protocol) which contains or references all of the information related to the SUT and its testing environment, which will enable the test laboratory to run an appropriate test suite against the SUT

SIP number: number conforming to the numbering and structure specified in ITU-T Recommendation E.164 [11]

System Under Test (SUT): real open system in which the SUT resides

user: access protocol entity at the User side of the user-network interface where a T reference point or coincident S and T reference point applies

3.2 Abbreviations

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

ATC	Abstract Test Case
ATM	Abstract Test Method
ATP	Access Transport Parameter
ATS	Abstract Test Suite
BCI	Backward Call Indicators
CPS	Calling Party's Category
DSS1	Digital Subscriber System No. 1
FCI	Forward Call Indicators
HLC	High Layer Compatibility
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
IUT	Implementation Under Test
MOT	Means Of Testing
NCI	Nature of Connection Indicators
OBCI	Optional Backward Call Indicators
OFCI	Optional Forward Call Indicator
PICS	Protocol Implementation Conformance Statement
PIXIT	Protocol Implementation eXtra Information for Testing
SUT	System Under Test
TMR	Transmission Medium Requirement
TP	Test Purpose
TSS	Test Suite Structure
TTCN	Tree and Tabular Combined Notation

NOTE: The ISUP message acronyms can be found in table 2/ ITU-T Recommendation Q.762 [3].

4 Implementation under test and test methods

4.1 Identification of the system and implementation under test

FFS

5 Test Suite Structure (TSS)

The Test Suite Structure is in close alignment with ES 283 027 [1].

5.1 Interworking from SIP to ISUP (outgoing call)

SIP -ISUP Basic call		
	Sending of the Initial address message (IAM)	101xxx
	Sending of the Subsequent address message (SAM)	102xxx
	Sending of COT	103xxx
	Receipt of the Address complete message (ACM)	104xxx
	Receipt of the Call progress message (CPG)	105xxx
	Receipt of the answer message (ANM)	106xxx
	Receipt of the Connect message (CON)	107xxx
	Receipt of the Release message (REL)	108xxx
	Autonomous release at I-MGCF	109xxx
	Receipt of the BYE, CANCEL message / sending of a REL message	110xxx
	Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	111xxx
	Receipt of the SUSPEND Message (SUS)	111xxx
	Receipt of the RESUME Message (RES)	112xxx

Figure 1: Basic call -
Test suite structure for interworking between SIP to ISUP (outgoing call)

5.2 Interworking from ISUP to SIP (incoming call)

ISUP-SIP Basic call		
	Sending of the INVITE message	301xxx
	Receipt of the Subsequent address message (SAM)	302xxx
	Sending of the Address complete message (ACM)	303xxx
	Sending of the Call progress message (CPG)	304xxx
	Sending of the answer message (ANM)	305xxx
	Sending of the Connect message (CON)	306xxx
	Receipt of the Release message (REL)	307xxx
	Sending of the Release Message (REL)	308xxx
	Autonomous release	309xxx
	Receipt of Reset circuit message (RSC)	310xxx
	Receipt of Circuit group reset message (GRS)	311xxx
	Receipt of Circuit group blocking message (CGB) with the indication hardware failure oriented	312xxx

Figure 2: Basic call -
Test suite structure for interworking between ISUP to SIP (incoming call)

5.3 Supplementary Services - Interworking from SIP to ISUP (outgoing call)

SIP-ISUP Supplementary Services		
	Calling Line Identification (CLI)	501xxx
	Call Hold (HOLD)	502xxx
	Terminal Portability (TP)	503xxx
	Conference Calling (CONF)	504xxx
	Three-Party (3PTY)	505xxx
	Connected Line Identification (COL)	506xxx
	Malicious call identification (MCID)	507xxx
	Subaddressing (SUB)	508xxx
	Call Diversion (CDIV)	509xxx
	Call Waiting (CW)	510xxx
	User to User Signalling (UUS)	511xxx
	Explicit Call transfer (ECT)	512xxx
	Completion of Call to Busy Subscriber (CCBS)	513xxx
	Completion of Calls on No reply (CCNR)	514xxx
	Anonymous Call Rejection (ACR)	515xxx
	Closed user group (CUG)	516xxx

Figure 3: Supplementary Services -
Test suite structure for interworking between SIP to ISUP (outgoing call)

5.4 Supplementary Services - Interworking from ISUP to SIP (incoming call)

ISUP-SIP		
	Calling Line Identification (CLI)	601xxx
	Call Hold (HOLD)	602xxx
	Terminal Portability (TP)	603xxx
	Conference Calling (CONF)	604xxx
	Three-Party (3PTY)	605xxx
	Connected Line Identification (COL)	606xxx
	Subaddressing (SUB)	607xxx
	Closed User Group (CUG)	608xxx
	Call Diversion (CDIV)	609xxx
	User to User Signalling (UUS)	610xxx
	Explicit Call transfer (ECT)	611xxx
	Anonymous Call Rejection (ACR)	612xxx
	Call waiting (CW)	613xxx
	Malicious call identification (MCID)	614xxx

Figure 4: Supplementary Services -
Test suite structure for interworking between ISUP to SIP (outgoing call)

6 Test purposes (TP)

6.1 Introduction

For each test requirement a Test Purpose (TP) is defined.

6.1.1 Test purpose (TP) naming convention

For each test requirement a Test Purpose (TP) is defined.

All test purposes belong to the main group ISUP_SIP_Interworking. Groups are organized according to the test suite structure (TSS). Each test purpose is presented in a separate table. The first row of the table contains the following items:

TP	Identifier of the test purpose;
SIP reference	the reference to the requirement in the DSS1 layer 3 Recommendation, which led to the TP;
ISUP reference	the reference to the requirement in the interworking specification and the requirement in the SIP-UP Recommendation, which led to the TP.

6.1.2 Source of test purpose definition

The test purposes have been developed based on ES 283 027 [1] as an endorsement of TS 129 163 [15].

6.1.3 Test purpose structure

The test purpose structure is according to the test suite structure (TSS).

6.2 Test purposes for the basic call

6.2.1 Interworking from SIP to ISUP (Outgoing Call)

6.2.1.1 Sending of the Initial Address Message (IAM)

<table>
<tr><th>TP101001</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
ES 283 027 [1], clause 7.2.3.1.1</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>Normal call setup without precondition reequirement</i>
Ensure that if the SIP precondition extension is not included in the Supported or Require header, the I-MGCF shall send an IAM immediately after the reception of the INVITE, The I-MGCF shall set the continuity indicators to "Continuity check not required".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments:</td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP101002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.1**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:		
Test purpose:	*Call setup with precontion tag in the Supported header and preconditions are fullfield successful* Ensure if a Continuity Check procedure is supported in the ISUP network and SIP precondition extension are included in the SIP Supported header and the preconditions are indicated as fullfield in the SDP, the I-MGCF shall send the IAM immediately after the reception of the INVITE. The preconditions met is sent in the 200 OK INVITE.	
SIP Parameter values:	INVITE: Supported: 100rel, precondition SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv 200 OK INVITE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	
ISUP Parameter values:	IAM: Continuity indicator: Continuity check not required	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.1**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	NOT PICS 4/4 AND NOT PICS 4/5	
ISUP selection criteria:		
Test purpose:	*Call setup with precontion tag in the Supported header and preconditions are fullfield unsuccessful* Ensure if the received SDP indicates that precondition is fulfilled the I-MGCF shall set the continuity indicators to "continuity check is not required". The SUT does not an answer to the precondition requirement.	
SIP Parameter values:	INVITE: Supported: 100rel, precondition SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv	
ISUP Parameter values:	IAM: Continuity indicator: Continuity check not required	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.1**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 4/1	
Test purpose:	*Call setup with precondition tag in the Require header and requirement for recource reservation* Ensure if the INVITE request contains the precondition tag in the Require header the received SDP indicates that precondition is not fulfilled the I-MGCF shall set the continuity indicators to "continuity check performed on a previous circuit" or "required on this circuit".	
SIP Parameter values:	INVITE: Require: precondition SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv 183: Require: 100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv a=conf:qos remote sendrecv UPDATE: SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv 200 OK UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	
ISUP Parameter values:	IAM: "continuity check required on this circuit" or "Continuity check performed on a previous circuit"	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE → → COT 200 OK UPDATE ← 180 Ringing ← ← ACM PRACK → 200 OK PRACK ← Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

<table>
<tr><th>TP101005</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.1</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 4/4 AND PICS 4/5</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 4/1</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Call setup with precondition tag in the Supported header and requirement for recource reservation</i>

Ensure if the INVITE request contains the precondition tag in the Supported header the received SDP indicates that precondition is not fulfilled the I-MGCF shall set the continuity indicators to "continuity check performed on a previous circuit" or "required on this circuit".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INVITE: Supported: precondition
SDP a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv

183: Require: 100rel
SDP a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv

UPDATE:
SDP a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv

200 OK UPDATE
SDP a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM: "continuity check required on this circuit" or "Continuity check performed on a previous circuit"</td></tr>
<tr><td>Comments:</td><td colspan="2">SIP SUT ISUP
INVITE → → IAM
183 Session Progress ←
PRACK →
200 OK PRACK ←

UPDATE → → COT
200 OK UPDATE ←

180 Ringing ← ← ACM
PRACK →
200 OK PRACK ←
Ringing tone
200 OK INVITE ← ← ANM
ACK →
Conversation
BYE → → REL
200 OK BYE ← ← RLC</td></tr>
</table>

TP101006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.1**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	NOT PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 4/1	
Test purpose:	*Call setup with precondition tag in the Require header and requirement for recource reservation* Ensure if the INVITE request contains the precondition tag in the Require header the received SDP indicates that precondition is not fulfilled the I-MGCF shall send a 5xx final provisional response if preconditions are not supported.	
SIP Parameter values:	INVITE: Require: precondition SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv	
ISUP Parameter values:		
Comments:	**SIP** SUT **ISUP** INVITE → 580 Precondition Failure ← ACK →	

TP101007	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.1 RFC 3264 [18]**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 1/3 AND NOT PICS 4/1	
Test purpose:	*COT procedure not supported, IAM delayed until preconditions met* Ensure if Continuity Check procedure is not supported in the ISUP network, and the SDP in the received INVITE request contains preconditions not met, the I-MGCF shall delay sending the IAM until the SIP preconditions are met and set the continuity indicators in the resulting IAM to "Continuity check not required".	
SIP Parameter values:	INVITE: Require: precondition SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv 183: Require: 100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv a=conf:qos remote sendrecv UPDATE: SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv 200 OK UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	
ISUP Parameter values:	IAM Continuity Indicator: continuity check required on this circuit, COT Continuity Indicator: continuity check successful;	
Comments:	**SIP** SUT **ISUP** INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE → → IAM 200 OK UPDATE ← 180 Ringing ← ← ACM PRACK → 200 OK PRACK ← Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101008	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.1** **RFC 3264 [18]**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 1/3 AND NOT PICS 1/4 AND PICS 4/1	
Test purpose:	*Media type not supported, call setup rejected* Ensure that the I-MGCF shall reject an INVITE request for a session only containing unsupported media types by sending a status code 488 "Not Acceptable Here.	
SIP Parameter values:	SDP: media type not supported in the SUT (PIXIT)	
ISUP Parameter values:		
Comments:	**SIP** SUT **ISUP** INVITE → 488 Not Acceptable Here ← ACK →	

TP101009	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.3.3.1.1** **RFC 3264 [18]**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*SUT rejects unsupported media types* Ensure that If several media streams are contained in a single INVITE request, the I-MGCF shall select one of the supported media streams, reserve the codec(s) for that media stream, and reject the other media streams and unselected codecs in the SDP answer, as detailed in RFC 3264 [18]. If supported audio media stream(s) and supported non-audio media stream(s) are contained in a single INVITE request, an audio stream should be selected.	
SIP Parameter values:	Offer: m=audio 4711 RTP/AVP 8 m= video 4713 RTP/AVP 31 Answer: m=audio 4711 RTP/AVP 8 m=video 0 RTP/AVP 31	
ISUP Parameter values:		
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101010	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.1** **RFC 3264 [18]**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/15	
ISUP selection criteria:		
Test purpose:	*To tag included in 183 provisional response* Ensure that The I-MGCF shall include a To tag in the first backward non-100 provisional response, in order to establish an early dialog as described in RFC 3261 [6]	
SIP Parameter values:	183 To taq included	
ISUP Parameter values:	ACM: oBCi "inband info available"	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 183 Session Progress ← ← ACM(no indication) 180 Ringing ← ← CPG(Alerting) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101011	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.1** **RFC 3264 [18]**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*To tag included in 180 provisional response* Ensure that The I-MGCF shall include a To tag in the first backward non-100 provisional response, in order to establish an early dialog as described in RFC 3261 [6]	
SIP Parameter values:	180 To tag included	
ISUP Parameter values:		
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL ~~200 OK BYE ← ← RLC~~	

TP101012	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clauses 7.2.3.1.2.2 and 7.2.3.1.2.3**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 2/1	
ISUP selection criteria:	PICS NOT 4/16	
Test purpose:	*Setting of nature of connection indicator and forward call indicator* Ensure that the SUT on receipt of an INVITE message: sends an IAM message, where the Calling party's category is set to "Ordinary calling subscriber", the **Nature of Connection Indicators (NCI)** encoded as follows: • Satellite indicator set to: One satellite circuit in the connection". • Echo control device indicator set to: "Outgoing echo control device included". • The **Forward call indicator** is encoded as follows: Interworking indicator: Interworking encounterd. ISUP/BICC Indicator: ISDN User part/BICC not used all the way. ISUP/BICC Preference indicator: ISDN user part/BICC not required all the way. ISDN access indicator: Originating acces non-ISDN.	
SIP Parameter values:		
ISUP Parameter values:	**Nature of Connection Indicators (NCI):** Satellite indicator set to: "One satellite circuit in the connection" Echo control device indicator set to: "Outgoing echo control device included" **Forward Call Indicators (FCI):** Interworking indicator: interworking encounterd ISDN user part indicator: ISDN user part/BICC not used all the way ISDN access indicator: originating access non-ISDN ISDN user part preference indicator: ISDN user part/BICC not required all the way	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101013	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.2.2** **Q.767**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Setting of nature of connection indicator and forward call indicator T38 codec received* Ensure that the SUT on receipt of an INVITE message with SDP m line T:38: sends an IAM message, where the Calling party's category is set to "Ordinary calling subscriber", the **Nature of Connection Indicators (NCI)** encoded as follows: • Satellite indicator set to: "One satellite circuit in the connection" . - Echo control device indicator set to: "Outgoing echo control device **not** included". - the **Forward call indicator** is encoded as follows: Interworking indicator: Interworking encounterd ISUP/BICC Indicator: ISDN User part/BICC not used all the way ISUP/BICC Preference indicator: ISDN user part/BICC not required all the way ISDN access indicator: Originating acces non-ISDN	
SIP Parameter values:	INVITE with SDP m line T:38	
ISUP Parameter values:	**Nature of Connection Indicators (NCI):** Satellite indicator set to: "One satellite circuit in the connection" Echo control device indicator set to: "Outgoing echo control device **not** included" **Forward Call Indicators (FCI):** Interworking indicator: interworking encounterd ISDN user part indicator: ISDN user part/BICC not used all the way ISDN access indicator: originating access non-ISDN ISDN user part preference indicator: ISDN user part/BICC not required all the way	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101014	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.2.3** **Q.767**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 2/3	
ISUP selection criteria:	PICS 4/16	
Test purpose:	*Setting of nature of connection indicator and forward call indicator indicating ISDN and TMR 64 kBit/s* Ensure that the SUT on receipt of an INVITE message with SDP m line CLEARMODE: • sends an IAM message, where the Calling party's category is set to "Ordinary calling subscriber", if the TMR = 64 kBit/s unrestricted is used the **Nature of Connection Indicators (NCI)** encoded as follows: - the **Nature of Connection Indicators (NCI)** encoded as follows: - Satellite indicator set to: "One satellite circuit in the connection" - Echo control device indicator set to: outgoing echo control device not included. - the **Forward call indicator** is encoded as follows: Interworking indicator: No interworking encounterd ISUP/BICC Indicator: ISDN User part/BICC used all the way ISUP/BICC Preference indicator: ISDN user part/BICC not required all the way ISDN access indicator: Originating acces ISDN.	
SIP Parameter values:		
ISUP Parameter values:	**Nature of Connection Indicators (NCI):** Satellite indicator set to: "One satellite circuit in the connection" Echo control device indicator set to: outgoing echo control device not included **Forward Call Indicators (FCI):** Interworking indicator: No interworking encounterd ISDN user part indicator: ISDN user part/BICC used all the way ISDN access indicator: originating access ISDN ISDN user part preference indicator: ISDN user part/BICC not required all the way	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

<table>
<tr><th>TP101015</th><th colspan="4">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
ES 283 027 [1], clause 7.2.3.1.2.5</th></tr>
<tr><td>TSS reference:</td><td colspan="6">SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="6">Based on table 1</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="6"></td></tr>
<tr><td>Test purpose:</td><td colspan="6"><i>Mapping of SDP into the TMR</i>
Ensure that the SUT in the Idle state on receipt of an INVITE message containing the media description defined in table 1 with the "a =" "b =" and "m=" lines set to a_b_m_LINE_VALUE:
sends an IAM message, with the Transmission Medium Requirement (TMR) parameter set to TMR_VALUE.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6">INVITE; a_b_m_LINE_VALUE</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="6">IAM; TMR: ISUP_TMR</td></tr>
<tr><td rowspan="9">Comments:</td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISUP</td><td></td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td><td></td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td><td></td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td><td></td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td><td></td></tr>
</table>

Table 1

Values for test purposes TP101015						
	a_b_m_LINE_VALUE					
m= line				**b= line**	**a= line**	**TMR_VALUE**
test purposes	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>	TMR codes
VA_01	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000	*"3,1KHz audio"*
VA_02	audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	*"3,1KHz audio"*
VA_03	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000	*"3,1KHz audio"*
VA_04	audio	RTP/AVP	Dynamic PT	AS: 64 kbit/s	rtpmap:<dynamic-PT> CLEARMODE/8000 (NOTE 2)	*"64 kbit/s unrestricted"*
VA_05	image	udptl	t38 [31]	N/A or up to 64 kbit/s	Based on ITU-T Recommendation T.38 [31]	*"3,1 kHz audio"*
VA_06	image	tcptl	t38 [31]	N/A or up to 64 kbit/s	Based on ITU-T Recommendation T.38 [31]	*"3,1 kHz audio"*
VA_07	audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	*"3,.1KHz audio"*

NOTE 1: In this table the codec G.711 is used only as an example. Other codecs are possible.

NOTE 2: CLEARMODE is specified in RFC 4040 [19].

NOTE 3: If the b=line indicates a bandwidth greater than 64 kbit/s then the call may use compression techniques or reject the call with a 415 response indicating that only one media stream of 64 kbit/s is supported.

NOTE 4: <bandwidth value> for <modifier> of AS is in units of kbit/s.

<table>
<tr><th>TP101016</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
ES 283 027 [1], clause 7.2.3.1.2.5</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 2/3</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5">PICS 4/18 AND PICS 4/19</td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>Fallback connection type supported: Mapping of the second PSTN XML BearerCapability elements into TMR and USI prime</i>
Ensure that when the INVITE request includes multiple PSTN XML bearer information element:
If the first stated codec in the INVITE is a codec appearing in table 1 and is the equivalent as stated within the second Bearer Capability in the XML Bearer Capability element then the I-MGCF shall map the XML Bearer Capability element into the TMR and USI prime and shall set the TMR to "64 kBit/s preferred".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">first BC: 3,1 kHz audio or speech
second BC: unrestricted digital information with tones/announcements</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5">USI Prime: unrestricted digital information with tones/announcements
TMR: 64 kBit/s preferred</td></tr>
<tr><td rowspan="10">Comments:</td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

<table>
<tr><th>TP101017</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
ES 283 027 [1], clause 7.2.3.1.2.5a</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 4/18</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5">PICS 4/19</td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>Fallback connection type supported: Mapping of the first PSTN XML BearerCapability elements into USI and TMR prime</i>
Ensure when the INVITE request includes multiple PSTN XML BearerCapability then the I-MGCF shall:
If the second stated codec in the INVITE is a codec appearing in table 1 and is the equivalent as stated within the first Bearer Capabililty in the XML Bearer Capability element then the I-MGCF shall map the XML Bearer Capability element into the TMR prime and USI and shall map the TMR prime from the PSTN XML BearerCapability (InformationTransferCabability).</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">first BC: 3,1 kHz audio or speech
second BC: unrestricted digital information with tones/announcements</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5">USI: 3,1 kHz audio or speech
TMR prime: 3,1 kHz audio or speech</td></tr>
<tr><td rowspan="10">Comments:</td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP101018	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.5a**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*Fallback supported. Discard the second PSTN XML BearerCapability element if it is not equivalent to the first codec in the SDP* Ensure when the INVITE request includes multiple PSTN XML BearerCapability then the I-MGCF shall: if the compared first codec stated within the INVITE is not equivalent as stated within the second XML Bearer Capability element, then the second XML Bearer Capability element shall be discarded.	
SIP Parameter values:	PSTN XML BC 1 (speech or 3,1 kHz audio) PSTN XML BC 2 (unrestricted digital information with tones and announcements) SDP: m =audio xxx, RTP/AVP 0 8	
ISUP Parameter values:	USI: not included TMR. 3,1 kHz audio	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101019	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.5**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of PSTN XML BearerCapability element into the **USI** parameter* Ensure that the SUT in the Idle state on receipt of an INVITE message containing the XML BearerCapability element the mapping of the USI ISUP_USI shall be taken from the PSTN XML BearerCapability value ISDN_BC.	
SIP Parameter values:	PSTN XML BearerCapability ISDN_BC	
ISUP Parameter values:	IAM: USI = ISUP_USI	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values and selection criteria for the test purpose TP101019		
VA_01	ISDN_BC = speech	ISUP_USI = speech
VA_02	ISDN_BC = 3,1 kHz audio	ISUP_USI = 3,1 kHz audio
VA_03	ISDN_BC = Unrestricted digital information	ISUP_USI = Unrestricted digital information

<table>
<tr><th>TP101020</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.2.5</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 4/18</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Mapping of PSTN XML BearerCapability element into the TMR parameter</i>
Ensure that the SUT in the Idle state on receipt of an INVITE message containing one PSTN XML BearerCapability value ISDN_BC_ITC
sends an IAM message, with the Transmission Medium Requirement (TMR) parameter set to ISUP_TMR.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INVITE; PSTN XML BearerCapability</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM: TMR</td></tr>
<tr><td>Comments:</td><td colspan="2">SIP SUT ISUP
INVITE → → IAM
180 Ringing ← ← ACM
Ringing tone
200 OK INVITE ← ← ANM
ACK →
Conversation
BYE → → REL
200 OK BYE ← ← RLC</td></tr>
</table>

<table>
<tr><th colspan="3">Values and selection criteria for the test purpose TP101020</th></tr>
<tr><td>VA_01</td><td>ISDN_BC_ITC = speech
ISDN_BC_ITR = 64 kbits/s</td><td>ISUP_TMR = speech</td></tr>
<tr><td>VA_02</td><td>ISDN_BC_ITC = 3,1 kHz audio
ISDN_BC_ITR = 64 kbits/s</td><td>ISUP_TMR = 3,1 kHz audio</td></tr>
<tr><td>VA_03</td><td>ISDN_BC_ITC = unrestricted digital information
ISDN_BC_ITR = 64 kbits/s</td><td>ISUP_TMR = 64 kbits/s unrestricted</td></tr>
</table>

<table>
<tr><th>TP101021</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.2.5</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 4/18</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>No PSTN XML received, mapping of HLC in ATP</i>
Ensure that the SUT in the Idle state on receipt of an INVITE message, with the media description defined in table 2 with the "a =" "b =" and "m=" lines to lines set to a_b_m_LINE_VALUE:
• sends an IAM message with the Access transport parameter containing the HLC information element.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INVITE: a_b_m_LINE_VALUE</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM; Access transport parameter HLC: HLC_VALUE</td></tr>
<tr><td>Comments:</td><td colspan="2">SIP SUT ISUP
INVITE → → IAM
180 Ringing ← ← ACM
Ringing tone
200 OK INVITE ← ← ANM
ACK →
Conversation
BYE → → REL
200 OK BYE ← ← RLC</td></tr>
</table>

Table 2

Values for test purposes TP101021						
M= line				**b= line**	**a= line**	**HLC parameter HLC_VALUE**
Test purposes	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>	HLC_VALUE
				see note 1		
VA_01	Audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	See note 2
VA_02	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000	See note 2
VA_03	Audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	See note 2
VA_04	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000	See note 2
VA_05	Image	Udptl	t38	N/A or up to 64 kbit/s	Based on ITU-T Recommendation T.38 [31]	*"Facsímile Group 2/3"*
VA_06	Image	Tcptl	t38	N/A or up to 64 kbit/s	Based on ITU-T Recommendation T.38 [31]	*"Facsímile Group 2/3"*
NOTE 1: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s. NOTE 2: HLC normally absent in this case. It is possible for HLC to be present with the value "Telephony", although clause 6.3.1/ITU-T Recommendation Q.939 [13] indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.						

TP101022	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.10**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of PSTN XML BearerCapability and HighLayerCompatibility in ATP HLC* Ensure that the SUT in the Idle state on receipt of a INVITE message with a PSTN XML BearerCapability BC_VALUE and the **PSTN XML HighLayerCompatibility** HLC_VALUE sends an IAM message with the **Access transport parameter** containing the received **PSTN XML HighLayerCompatibility**].	
SIP Parameter values:	INVITE PSTN XML BearerCapability: **BC_VALUE** PSTN XML HighLayerCompatibility: **HLC_VALUE**	
ISUP Parameter values:	IAM; **Access transport parameter HLC:** HLC_VALUE	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values and selection criteria for the test purpose TP101022	
VA_01	HLC_VALUE = Telephony BC_VALUE = speech
VA_02	HLC_VALUE = Facsimile Group 2/3 (ITU-T Recommendation F.182 [20]) BC_VALUE = 3,1 kHz audio
VA_03	HLC_VALUE == Facsimile Group 4 Class I (ITU-T Recommendation F.184 [20]) BC_VALUE = Unrestricted digital information
VA_04	HLC_VALUE == Teletex service, basic and mixed mode of operation (ITU-T Recommendation F.230 [22]) and facsimile service Group 4, Classes II and III (ITU-T Recommendation F.184 [21]) BC_VALUE = Unrestricted digital information
VA_05	HLC_VALUE == Teletex service, basic and processable mode of operation (ITU-T Recommendation F.220 [23]) BC_VALUE = Unrestricted digital information
VA_06	HLC_VALUE = Teletex service, basic mode of operation (ITU-T Recommendation F.200 [24]) BC_VALUE = Unrestricted digital information
VA_07	HLC_VALUE = Syntax based Videotex (ITU-T Recommendations F.300 [25] and T.102) BC_VALUE = Unrestricted digital information
VA_08	HLC_VALUE = International Videotex interworking via gateways or interworking units (ITU-T Recommendations F.300 [25] and T.101) BC_VALUE = Unrestricted digital information
VA_09	HLC_VALUE = Telex service (ITU-T Recommendation F.60 [26]) BC_VALUE = Unrestricted digital information
VA_10	HLC_VALUE = Message Handling Systems (MHS) (X.400 - Series Recommendations) BC_VALUE = Unrestricted digital information
VA_11	HLC_VALUE = OSI application (X.200 - Series ITU-T Recommendations) BC_VALUE = Unrestricted digital information
VA_12	HLC_VALUE = Audio visual (ITU-T Recommendation F.721 [27]) BC_VALUE = Unrestricted digital information

TP101023	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.10**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of PSTN XML BearerCapability and HighLayerCompatibility in User Teleservice parameter* Ensure that the SUT in the Idle state on receipt of a INVITE message with **PSTN XML BearerCapability** BC_VALUE and the **PSTN XML HighLayerCompatibility** HLC_VALUE sends an IAM message with the User Teleservice parameter containing the received **PSTN XML HighLayerCompatibility**.	
SIP Parameter values:	INVITE **PSTN XML BearerCapability: BC_VALUE** **PSTN XML HighLayerCompatibility:** HLC_VALUE	
ISUP Parameter values:	IAM; User teleservice parameter	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values and selection criteria for the test purpose TP101023	
VA_01	HLC_VALUE = Telephony BC_VALUE = speech
VA_02	HLC_VALUE = Facsimile Group 2/3 (ITU-T Recommendation F.182 [20]) BC_VALUE = 3,1 kHz audio
VA_03	HLC_VALUE == Facsimile Group 4 Class I (ITU-T Recommendation F.184 [21]) BC_VALUE = Unrestricted digital information
VA_04	HLC_VALUE == Teletex service, basic and mixed mode of operation (ITU-T Recommendation F.230 [22]) and facsimile service Group 4, Classes II and III (ITU-T Recommendation F.184 [21]) BC_VALUE = Unrestricted digital information
VA_05	HLC_VALUE == Teletex service, basic and processable mode of operation (ITU-T Recommendation F.220 [23]) BC_VALUE = Unrestricted digital information
VA_06	HLC_VALUE = Teletex service, basic mode of operation (ITU-T Recommendation F.200 [24]) BC_VALUE = Unrestricted digital information
VA_07	HLC_VALUE = Syntax based Videotex (ITU-T Recommendations F.300 [25] and T.102) BC_VALUE = Unrestricted digital information
VA_08	HLC_VALUE = International Videotex interworking via gateways or interworking units (ITU-T Recommendations F.300 [25] and T.101) BC_VALUE = Unrestricted digital information
VA_09	HLC_VALUE = Telex service (ITU-T Recommendation F.60 [26]) BC_VALUE = Unrestricted digital information
VA_10	HLC_VALUE = Message Handling Systems (MHS) (X.400 - Series Recommendations) BC_VALUE = Unrestricted digital information
VA_11	HLC_VALUE = OSI application (X.200 - Series Recommendations) BC_VALUE = Unrestricted digital information
VA_12	HLC_VALUE = Audio visual (ITU-T Recommendation F.721 [27]) BC_VALUE = Unrestricted digital information

TP101024	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.10**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of PSTN XML LowLayerCompatibility in ATP LLC* Ensure that the SUT in the Idle state on receipt of a INVITE message with **a PSTN XML LowLayerCompatibility** sends an IAM message with the **Access transport parameter** containing the **PSTN XML LowLayerCompatibility** as received in the INVITE message.	
SIP Parameter values:	INVITE ; **PSTN XML LowLayer Compatibility**: **LLC_VALUE** (PIXIT)	
ISUP Parameter values:	IAM; Access transport parameter LLC: **LLC_VALUE** (PIXIT)	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101025	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.10**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of PSTN XML ProgressIndicator in ATP PI* Ensure that the SUT in the Idle state on receipt of a INVITE message with a valid **PSTN XML ProgressIndicator** sends an IAM message with the **Access transport parameter** containing the **PSTN XML ProgressIndicator** as received in the INVITE message.	
SIP Parameter values:	INVITE; **PSTN XML ProgressIndicator**: PI_VALUE	
ISUP Parameter values:	IAM; **progress indicator** PI_VALUE	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values and selection criteria for the test purpose TP101025	
VA_01	PI_VALUE = Call is not end-to-end ISDN; further call progress information is available in-band (# 1)
VA_02	PI_VALUE = Originating access is non ISDN (#3)

<table>
<tr><th>TP101026</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.2.9</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 4/3</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>HOP counter derived from the Max-Forward header</i>

Ensure that the SUT the I-MGCF shall derive the Hop Counter parameter value from the Max-Forwards header field value by applying a factor. The Hop Counter for a given message should never increase and should decrease by at least 1 with each successive visit to an MGCF, regardless of interworking, and similarly for Max-Forwards in the SIP domain.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Max-Forward header</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM: Hop Counter parameter value</td></tr>
<tr><td rowspan="2">Comments:</td><td colspan="2">SIP SUT ISUP
INVITE → → IAM
180 Ringing ← ← ACM
Ringing tone
200 OK INVITE ← ← ANM
ACK →
Conversation
BYE → → REL
200 OK BYE ← ← RLC</td></tr>
<tr><td colspan="2">The initial and successively mapped values of Hop Counter should be large enough to accommodate the maximum number of hops that might be expected of a validly routed call.</td></tr>
</table>

<table>
<tr><th>TP101027</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.2.1</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">NOT PICS 1/5</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Mapping of Request URI into called party number "national number"</i>

Analyse the information contained in received Request URI E.164 address.
If CC is country code of the network in which the next hop terminates, then set Nature of Address indicator to "National (significant) number". The country code is removed from the numberstring.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INVITE: Request URI</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM: Called party number</td></tr>
<tr><td>Comments:</td><td colspan="2">SIP SUT ISUP
INVITE → → IAM
180 Ringing ← ← ACM
Ringing tone
200 OK INVITE ← ← ANM
ACK →
Conversation
BYE → → REL
200 OK BYE ← ← RLC</td></tr>
</table>

TP101028	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.1**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:	PICS 1/5	
Test purpose:	*Mapping of Request URI into called party number "international number"* Analyse the information contained in received Request URI E.164 address. If CC is not the country code of the network in which the next hop terminates, then set Nature of Address indicator to "**International number**".	
SIP Parameter values:	INVITE: Request URI	
ISUP Parameter values:	IAM: Called party number	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101029	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.1**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of Request URI into called party number. Setting of Numbering plan and internal network number indicator* Ensure that the SUT on receipt of an INVITE message with a Called party number contained in the userinfo component of the Request-URI: • Internal Network Number Indicator: routing to internal network number not allowed. • Numbering plan Indicator: 001 ISDN (Telephony) numbering plan.	
SIP Parameter values:	INVITE: Request URI	
ISUP Parameter values:	IAM: Called party number	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101030	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.1**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of number digits in the Requesr URI into address signals in the Called party number CC is the same as the MGCF is located* Analyse the information contained in received E.164 address. If CC is country code of the network in which the next hop terminates, then remove "+CC" and use the remaining digits to fill the Address signals.	
SIP Parameter values:	INVITE: Request URI	
ISUP Parameter values:	IAM: Called party number address signals	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101031	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.1**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of number digits in the Requesr URI into address signals in the Called party number CC is not the same as the MGCF is located* Analyse the information contained in received E.164 address. If CC is not the country code of the network in which the next hop terminates, then remove "+" and use the remaining digits to fill the Address signals.	
SIP Parameter values:	INVITE: Request URI	
ISUP Parameter values:	IAM: Called party number address signals	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP101032	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.1**
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of calling party category* Ensure that a cpc SIP_CPC parameter SIP_CPC received in the P-Asserted-Identity and the "language" in the Accept-Contact SIP_LANG header is mapped into the calling party parameter ISUP_CPC in the sent IAM	
SIP Parameter values:	INVITE: P-Asserted-Identity, Accept-Contact	
ISUP Parameter values:	IAM: Calling Party Category	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values for test purposes TP101032		
SIP_CPC		**ISUP_CPC**
cpc received in a P-Asserted-Identity	**Accept-Contact 'language' SIP_LANG**	**Sent Calling party's category**
operator	French	operator, language French
operator	English	operator, language English
operator	German	operator, language German
operator	Russian	operator, language Russian
operator	Spanish	operator, language Spanish
ordinary		ordinary calling subscriber
test		Test call
payphone		Payphone
cellular		mobile terminal located in the home PLMN
cellular-roaming		mobile terminal located in a visited PLMN
ieps		IEPS call marking for preferential call set up

6.2.1.2 Overlap procedure at the I-MGCF

FFS

6.2.1.3 Sending of COT

TP103001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.3**
TSS reference:	SIP-ISUP/Basic call/ COT	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 1/5 AND PICS 4/1	
Test purpose:	*COT is sent after precondition met* If the IAM has already been sent, the Continuity message shall be sent indicating "continuity check successful", when all of the following conditions have been met: - The requested preconditions (if any) in the IMS network have been met. - A possible outstanding continuity check procedure is successfully performed on the outgoing circuit.	
SIP Parameter values:	INVITE: Require: precondition SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv 183: Require: 100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv a=conf:qos remote sendrecv UPDATE: SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv 200 OK UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	
ISUP Parameter values:	IAM: "Continuity check performed on a previous circuit" or "Continuity check required on this circuit" COT continuity indicator: Continuity check successful;	
Comments:	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE → → COT 200 OK UPDATE ← 180 Ringing ← ← ACM PRACK → 200 OK PRACK ← Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

6.2.1.4 Receipt of the Address Complete Message (ACM)

TP104001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:		
Test purpose:	*Early ACM not interworked* Ensure that the SUT on receipt of an ACM message where the Called party status indicator is set to "no indication": • the ACM is not interworked.	
SIP Parameter values:		
ISUP Parameter values:	ACM Called party status: no indication;	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:		
Test purpose:	*Early ACM not interworked. Announcement provided by the terminating network* Ensure that the SUT on receipt of an ACM message where the Called party status indicator is set to "**no indication**"and during the establishment of the communication the PSTN/ISDN provides an announcement: • the ACM is not interworked.	
SIP Parameter values:		
ISUP Parameter values:	ACM Called party status: no indication;	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) Tones or announcement 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4A**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 4/15 AND PICS 2/1	
ISUP selection criteria:		
Test purpose:	*Early ACM terminating access ISDN and OBCI "inband info available" received. Sending of 183 containing a P-Early-Media header* Ensure that SUT on receipt of an ACM message where the **CPS indicator** is set to "no indication" the **ISUP indicator** is set to "ISUP is used all the way", the **ISDN access indicator** is set to "ISDN" and the **OBCI** with the in-band information is set to "Yes" and if the I-MGCF has received the P-Early-Media header in the INVITE request, and has not already sent a provisional response including a P-Early-Media header with parameters indicating authorization of early media, then the I-MGCF shall: • send the 183 Session Progress response with a P-Early-Media header authorizing early media.	
SIP Parameter values:	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8 183 Session Progress: P-Early-Media header	
ISUP Parameter values:	ACM; **CPS indicator:** no indication, **ISUP indicator:** ISUP is used all the way **ISDN access indicator:** ISDN **OBCI** in-band information: Yes	
Comments:	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← ← ACM (no indication) 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4A**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 4/15 AND PICS 2/1	
ISUP selection criteria:		
Test purpose:	*Early ACM BCI "*ISDN User Part not used all the way*" received. Sending of 183 containing a P-Early-Media header* Ensure that SUT on receipt of an ACM message where the **CPS indicator** is set to "no indication", the **ISUP indicator** is set to "ISUP is used not all the way", and if the I-MGCF has received the P-Early-Media header in the INVITE request, and has not already sent a provisional response including a P-Early-Media header with parameters indicating authorization of early media, then the I-MGCF shall: • send the 183 Session Progress response with P-Early Media and a P-Early-Media header authorizing early media.	
SIP Parameter values:	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8 183 Session Progress: P-Early-Media header	
ISUP Parameter values:	ACM; **CPS indicator**: no indication, **ISUP indicator**: ISUP is not used all the way" **OBCI** with the in-band information: Yes	
Comments:	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← ← ACM (no indication) 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4A**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 2/1 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Early ACM BCI "Terminating access non-ISDN" received. Sending of 183 containing a PSTN XML ProgressIndicator #2* Ensure that the SUT, on receipt of an ACM message where the **CPS indicator** is set to "no indication", the **ISUP indicator** is set to "ISUP is used all the way", the **ISDN access indicator** is set to "non-ISDN", then the I-MGCF shall: • send the 183 Session Progress response with **PSTN XML ProgressIndicator** body containing the progress descriptions "destination address is non-ISDN (#2)".	
SIP Parameter values:	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8 183 Session Progress: PSTM XML ProgressIndicator	
ISUP Parameter values:	ACM: **CPS indicator:** no indication, **ISUP indicator:** ISUP is used all the way **ISDN access indicator:** non-ISDN **OBCI** with the in-band information: No	
Comments:	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← ← ACM (no indication) 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4A**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 2/1 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Early ACM BCi "ISDN User Part not used all the way" received. Sending of 183 containing a PSTN XML ProgressIndicator #1* Ensure that the SUT on receipt of an ACM message where the **CPS indicator** is set to "no indication", the **ISUP indicator** is set to "ISUP not used all the way", then the I-MGCF shall: • send the 183 Session Progress response PSTN XML body containing the progress descriptions "call is not end-to-end ISDN, further call progress information is available in-band (#1).	
SIP Parameter values:	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8 183 Session Progress: P-Early Media and **PSTN XML ProgressIndicator** body containing the progress descriptions "call is not end-to-end ISDN, further call progress information is available in-band (#1)	
ISUP Parameter values:	ACM; **CPS indicator**: no indication, **ISUP indicator**: ISUP not used all the way	
Comments:	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← ← ACM (no indication) 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104007	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4A**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 2/1 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Early ACM OBCI "Inband-info available" received. Sending of 183 containing a PSTN XML ProgressIndicator #8* Ensure that the SUT on receipt of an ACM message where the **CPS indicator** is set to "no indication", the **OBCI** is set to "Inband-info available" then the I-MGCF shall: • send the 183 Session Progress response PSTN XML body containing the progress descriptions " in-band information or an appropriate pattern is now available" (#8).	
SIP Parameter values:	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8 183 Session Progress: P-Early Media and **PSTN XML ProgressIndicator** body containing the progress descriptions "in-band information or an appropriate pattern is now available" (#8)	
ISUP Parameter values:	ACM; **CPS indicator**: no indication, **OBCI**: in-band information or an appropriate pattern is now available	
Comments:	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← ← ACM (no indication) 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104008	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4A**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 2/1 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Early ACM terminating access ISDN and ATP with PI received. Sending of 183 containing a PSTN XML ProgressIndicator* Ensure that the SUT on receipt of an ACM message where the **CPS indicator** is set to "no indication", the **ISUP indicator** is set to "ISUP used all the way", the **ISDN access indicator** set to "ISDN", the **Access Transport Parameter** (ATP) containing the **progress indicator** set to PI_VALUE: • sends a 183 Session Progress message with **PSTN XML ProgressIndicator** set to PI_VALUE".	
SIP Parameter values:	183 Session Progress message **PSTN XML ProgressIndicator** set to PI_VALUE,	
ISUP Parameter values:	ACM, **CPS indicator:** no indication (00) **Called party"s category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: no interworking encountered (0) **ISUP indicator**: ISUP used all the way **ISDN access indicator:** ISDN **ATP progress indicator**: PI_VALUE **access delivery information**: Set-up message generated (IF PRESENT)	
Comments:	**SIP** SUT **ISUP** INVITE → → IAM 183 Session Progress ← ← ACM (no indication, ATP) 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104009	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4A**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 2/1 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Early ACM received, mapping of BCI into PSTN XML ProgressIndicator #7 in the sent 183* Ensure that the SUT on receipt of an ACM message where the **CPS indicator** is set to "no indication*"*, the **interworking indicator** I set to no interworking encountered, the **ISUP indicator** is set to "ISUP is used all the way", the **ISDN access indicator** is set to "ISDN" and the: • send the 183 Session Progress response with **PSTN XML ProgressIndicator** the progress descriptions "*Terminating access ISDN"(#7).*	
SIP Parameter values:	183 Session Progress message **PSTN XML ProgressIndicator** set to value #7	
ISUP Parameter values:	ACM: **CPS indicator:** no indication **interworking indicator**: no interworking encountered **ISUP indicator**: ISUP used all the way **ISDN access indicator:** terminating access ISDN	
Comments:	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← ← ACM (no indication) 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104010	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*ACM subscriber free received, a 180 is sent* Ensure that the SUT on receipt of an ACM message where the Called party status indicator is set to "subscriber free" where the ISUP indicator parameter set to **ISUP_ID**, the ISDN access indicator set to **ISDN_ACCES_ID** and the OBCI in-band information set to **OBCI_INBAND** then: • the **180 Ringing** SIP response is sent. Ensure that the in-band information can be transmitted to the calling user.	
SIP Parameter values:		
ISUP Parameter values:	ACM FCI: ISUP_ID, ISDN_ACCESS_ID OBCI: OBCI_INBAND;	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 3

test purposes	ISUP Parameter values:
VA_01	ACM ISUP_ID: ISUP not used all the way OBCI_INBAND: no
VA_02	ACM ISUP_ID: ISUP not used all the way OBCI_INBAND: yes
VA_03	ACM ISUP_ID: ISUP used all the way ISDN_ACCES_ID: non ISDN OBCI_INBAND: no
VA_04	ACM ISUP_ID: ISUP used all the way ISDN_ACCES_ID: non ISDN OBCI_INBAND: yes
VA_05	ACM ISUP_ID: ISUP used all the way ISDN access indicator: ISDN OBCI_INBAND: yes

TP104011	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4 ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 2/1 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ACM subscriber free received, mappimg of BCI into PSTN XML ProgressIndicator #7 in the sent 180* Ensure that the SUT on receipt of an ACM message where the **CPS indicator** is set to "subscriber free", the **ISUP indicator** is set to "ISUP is used all the way", the **ISDN access indicator** is set to "ISDN": • sends an 180 Ringing response with P-Early Media and **PSTN XML ProgressIndicator** "*Terminating access ISDN"(#7)*.	
SIP Parameter values:	180 Ringing response with P-Early Media and **PSTN XML ProgressIndicator** "*Terminating access ISDN"(#7)*.	
ISUP Parameter values:	ACM; **CPS indicator**: subscriber free **ISUP indicator**: ISUP is used all the way, **ISDN access indicator**: ISDN	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104012	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4** **ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 2/1 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ACM subscriber free and OBCI inband info available received, mappimg into PSTN XML ProgressIndicator(s) in the sent 180* Ensure that the SUT, having received the ACM message, where the **CPS indicator** is set to "subscriber free" where the **ISUP indicator parameter** set to ISUP_ID, the **ISDN access indicator** set to ISDN_ACCES_ID and the OBCI in-band information set to OBCI_INBAND: • sends an 180 Ringing response with PSTN XML body containing the progress descriptions set to PI_ID.	
SIP Parameter values:	180 Ringing P-Early Media and **PSTN XML ProgressIndicator** set to PI_ID.	
ISUP Parameter values:	ACM; **CPS indicator**: subscriber free **ISUP indicator**: ISUP_ID **ISDN access indicator**: ISDN_ACCES_ID	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values for test purposes TP104012		
test purposes	**ISUP Parameter values:**	**PSTN XML progress descriptions:**
VA_01	**ACM** ISUP_ID: ISUP not used all the way OBCI_INBAND: no	PI_ID: Call is not end-to-end ISDN (#1)
VA_02	**ACM** ISUP_ID: ISUP not used all the way OBCI_INBAND: yes	PI_ID: Call is not end-to-end ISDN (#1) and In-band information or appropriate pattern now available (#8)
VA_03	**ACM** ISUP_ID: ISUP used all the way ISDN_ACCES_ID: non ISDN OBCI_INBAND: no	PI_ID: Destination address is non-ISDN (#2)
VA_04	**ACM** ISUP_ID: ISUP used all the way ISDN_ACCES_ID: non ISDN OBCI_INBAND: yes	PI_ID: Destination address is non-ISDN (#2) and In-band information or appropriate pattern now available (#8)
VA_05	**ACM** ISUP_ID: ISUP used all the way ISDN access indicator: ISDN OBCI_INBAND: yes	PI_ID: In-band information or appropriate pattern now available (#8) and *Terminating access ISDN"(#7)*

TP104013	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4.1 ETS 300 356-1 [28] clauses 2.1.4, 2.2**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*ACM received containing a TMU parameter and ATP received, mapping into PSTN XML ProgressIndicators in the 180* Ensure that the SUT on receipt of an ACM message where the CPS indicator is set to "subscriber free", the ISUP indicator is set to "ISUP is used all the way", the ISDN access indicator is set to "ISDN" and the **Transmission Medium Used (TMU)** is included with the value ISDN_BC_VALUE and the **Access Transport Parameter (ATP)** set to ATP_VALUE: • sends an 180 Ringing message with a **PSTN XML BearerCapability** encoded BC_VALUE and with **PSTN XML ProgressIndicator** body containing the progress descriptions "*Terminating access ISDN"(#7).*	
SIP Parameter values:	INVITE; **PSTN XML first Bearer Capability**: INVITE _BC1 **PSTN XML second Bearer Capability**: INVITE _BC2 180 Ringing PSTN XML BC: BC_VALUE	
ISUP Parameter values:	ACM; CPS indicator: subscriber free, ISUP indicator: ISUP is used all the way ISDN access indicator: ISDN TMU: TMU_VALUE ATP: BC ATP_VALUE	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values and selection criteria for test purpose TP104013			
Test purposes	**ACM Parameter values**	**180 Ringing Parameter values:**	**INVITE parameter value**
VA_01	TMU_VALUE: speech ATP_VALUE: no BC	PSTN XML BC_VALUE: speech	PSTN XML INVITE BC1: speech INVITE BC2: unrestricted digital information with tones and announcements
VA_02	TMU_VALUE: 3,1 kHz ATP_VALUE: no BC	PSTN XML: BC_VALUE: 3,1 kHz audio	PSTN XML INVITE BC1: 3,1 kHz audio INVITE BC2: unrestricted digital information with tones and announcements
VA_03	TMU_VALUE: speech ATP_VALUE: BC = 3,1 kHz audio	PSTN XML BC_VALUE: 3,1 kHz audio	PSTN XML INVITE BC1: 3,1 kHz audio INVITE BC2: unrestricted digital information with tones and announcements
VA_04	TMU_VALUE: speech ATP_VALUE: BC = speech	PSTN XML BC_VALUE: speech	PSTN XML INVITE BC1: speech INVITE BC2: unrestricted digital information with tones and announcements
VA_05	TMU_VALUE: 3,1 kHz ATP_VALUE: BC = speech	PSTN XML BC_VALUE: speech	PSTN XML INVITE BC1: speech INVITE BC2: unrestricted digital information with tones and announcements
VA_06	TMU_VALUE: 3,1 kHz audio ATP_VALUE: BC = 3,1 kHz audio	PSTN XML BC_VALUE: 3,1 kHz audio	PSTN XML INVITE BC1: 3,1 kHz audio INVITE BC2: unrestricted digital information with tones and announcements

TP104014	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4.1** **ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*Fallback occurs in the early ACM mapping of TMU parameter and BC in ATP into 183 PSTN XML ProgressIndicators* Ensure that the SUT on receipt of an ACM message where the CPS indicator is set to "no indication", the ISUP indicator is set to "ISUP is used all the way", the ISDN access indicator is set to "ISDN" and the Transmission Medium Used (TMU) is included with the value TMU_VALUE and theBC in the Access Transport Parameter (ATP) set to ATP_VALUE: • sends a 183 Session Progress message and with the **PSTN XML BearerCapability** encoded ISDN_BC_VALUE and with **PSTN XML ProgressIndicator** body containing the progress descriptions "*Terminating access ISDN"(#7).*	
SIP Parameter values:	INVITE; **PSTN XML first Bearer Capability**: INVITE _BC1 **PSTN XML second Bearer Capability**: INVITE _BC2 183 Session Progress; **PSTN XML BearerCapability**: ISDN_BC_VALUE	
ISUP Parameter values:	ACM; CPS indicator: no indication, ISUP indicator: ISUP is used all the way ISDN access indicator: ISDN TMU: TMU_VALUE ATP: BC ATP_VALUE	
Comments:	SIP INVITE → 183 Session Progress ← 180 Ringing ← Ringing tone 200 OK INVITE ← ACK → Conversation BYE → 200 OK BYE ←	SUT → ISUP → IAM ← ACM ← CPG ← ANM → REL ← RLC

Values and selection criteria for test purpose TP104014			
Test purposes	**ACM Parameter values**	**183 Session Progress Parameter values:**	**INVITE parameter value**
VA_01	TMU_VALUE: speech ATP_VALUE: no BC	PSTN XML BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements
VA_02	TMU_VALUE: 3,1 kHz ATP_VALUE: no BC	PSTN XML BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements
VA_03	TMU_VALUE: speech ATP_VALUE: BC = 3,1 kHz	PSTN XML BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements
VA_04	TMU_VALUE: speech ATP_VALUE: BC = speech	PSTN XML BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements
VA_05	TMU_VALUE: 3,1 kHz ATP_VALUE: BC = speech	PSTN XML BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements
VA_06	TMU_VALUE: 3,1 kHz audio ATP_VALUE: BC = 3,1 kHz audio	PSTN XML BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements

<table>
<tr><th>TP104015</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.4A
ETS 300 356-1 [28] clause 2.1.4</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 4/18</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 4/19</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Fallback occurs in the early ACM mapping of HLC und BC in ATP into 183 PSTN XML ProgressIndicators</i>

Ensure that the SUT on receipt of an ACM message where the CPS indicator is set to "no indication", the ISUP indicator is set to "ISUP is used all the way", the ISDN access indicator is set to "ISDN" and containing an Access Transport Parameter (ATP) including a High Layer Compatibility (HLC) and containing the progress indicator #5: "interworking has occurred and has resulted in a telecommunication service change":

• sends a 183 Session Progress message with PSTN XML ProgressIndicator with the progress indication "interworking has occurred and has resulted in a telecommunication service change" (#5) "<i>Terminating access ISDN"(#7) and</i> with the PSTN XML HighLayerCapability.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INVITE: HLC: HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT)
180 Session Progress; PSTN XML ProgressIndicator: interworking has occurred and has resulted in a telecommunication service change (#5)
PSTN XML HighLayerCapability: HLC_VALUE2 (PIXIT)</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">ACM; CPS indicator: no indication,
ISUP indicator: ISUP is used all the way
ISDN access indicator: ISDN
ATP: progress indicator: interworking has occurred and has resulted in a telecommunication service change (#5)
HLC: HLC_VALUE2 (PIXIT)</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><td>SIP</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>CPG</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>
</td></tr>
</table>

TP104016	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4** **ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*ACM received, mapping of ATP HLC and PI #5 into 180 PSTN XML ProgressIndicators* Ensure that the SUT on receipt of an ACM message where the **CPS indicator** is set to "subscriber free", the **ISUP indicator** is set to "ISUP is used all the way", the **ISDN access indicator** is set to "ISDN" and containing an Access Transport Parameter (ATP) including a **High Layer Compatibility** (HLC) and containing the **progress indicator** #5: "interworking has occurred and has resulted in a telecommunication service change": • sends an 180 Ringing message with **PSTN XML ProgressIndicator** with the progress indication "interworking has occurred and has resulted in a telecommunication service change" (#5) "*Terminating access ISDN"(#7)* and with the **PSTN XML HighLayerCapability.**	
SIP Parameter values:	INVITE: **PSTN XML HighLayerCapability**: HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT) 180 Ringing; **PSTN XML ProgressIndicator**: interworking has occurred and has resulted in a telecommunication service change (#5) **PSTN XML HighLayerCapability**.: HLC_VALUE2 (PIXIT)	
ISUP Parameter values:	ACM; **CPS indicator**: subscriber free, **ISUP indicator**: ISUP is used all the way **ISDN access indicator**: ISDN **ATP: progress indicator**: interworking has occurred and has resulted in a telecommunication service change (#5) **HLC**: HLC_VALUE2 (PIXIT)	

Comments:	SIP		SUT		ISUP
	INVITE	→		→	IAM
	180 Ringing	←		←	ACM
			Ringing tone		
	200 OK INVITE	←		←	ANM
	ACK	→			
			Conversation		
	BYE	→		→	REL
	200 OK BYE	←		←	RLC

TP104017	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4** **ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 4/15 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ACM free terminating access ISDN received. Sending of 180 containing a P-Early-Media header* Ensure that SUT on receipt of an ACM message where the **CPS indicator** is set to "subscriber free" and if the I-MGCF has received the P-Early-Media header in the INVITE request, and has not already sent a provisional response including a P-Early-Media header with parameters indicating authorization of early media, then the I-MGCF shall: • send the 180 Ringing response with a P-Early-Media header authorizing early media	
SIP Parameter values:	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8 180 Ringing: P-Early-Media header	
ISUP Parameter values:	ACM; **CPS indicator**: free,	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104018	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4** **ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ACM free received contains an ATP conveying the LLC, mapping into the PSTN XML LLC in the sent 180* Ensure that the SUT on receipt of an ACM message containing the LLC parameter in the **ATP** set to LLC_VALUE: • sends 180 response with a **PSTN XML LowLayerCompatibility** information element set to LLC_VALUE	
SIP Parameter values:	180: **PSTN XML LowLayerCompatibility:** LLC_VALUE (PIXIT)	
ISUP Parameter values:	ACM; **ATP LLC**: LLC_VALUE	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP104019	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4** **ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ACM no indication received contains an ATP conveying the LLC, mapping into the PSTN XML LLC in the sent 183* Ensure that the SUT on receipt of an ACM message containing the LLC parameter in the **ATP** set to LLC_VALUE: • sends 183 Session Progress response with a **PSTN XML LowLayerCompatibility** information element set to LLC_VALUE	
SIP Parameter values:	180: **PSTN XML LowLayerCompatibility:** LLC_VALUE (PIXIT)	
ISUP Parameter values:	ACM; **ATP LLC**: LLC_VALUE	
Comments:	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← ← ACM 180 Ringing ← ← CPG Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

6.2.1.5 Receipt of the Call progress message (CPG)

TP105001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:		
Test purpose:	*CPG Alerting is mapped into a 180 ringing* Ensure that the SUT, having received the ACM message called party status indicator "no indication", on receipt of a CPG message where the **event information parameter event indicator** is set to "Alerting": • the 180 Ringing SIP response is sent.	
SIP Parameter values:		
ISUP Parameter values:	ACM: Called party status "no indication" CPG; **event information parameter event indicator**: Alerting	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) 180 Ringing ← ← CPG (Alerting) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP105002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:		
Test purpose:	*CPG Progress is not interworked* Ensure that the SUT, having received the ACM message called party status indicator "no indication", on receipt of a CPG message where the **event information parameter event indicator** is set to "Progress*"*: • the CPG is not interworked.	
SIP Parameter values:		
ISUP Parameter values:	ACM: Called party status "no indication" CPG; **event information parameter event indicator**: Progress	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) ← CPG (Progress) 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP105003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:		
Test purpose:	*CPG "in-band information or an appropriate pattern is now available" is not interworked* Ensure that the SUT, having received the ACM message called party status indicator "no indication", on receipt of a CPG message where the **event information parameter event indicator** is set to *"in-band information or an appropriate pattern is now available"*: • the CPG is not interworked.	
SIP Parameter values:		
ISUP Parameter values:	ACM: Called party status "no indication" CPG; **event information parameter event indicator**: in-band-information or an appropriate pattern is now available	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) ← CPG (Inband info) 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP105004	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4A**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	PICS 4/15	
ISUP selection criteria:		
Test purpose:	*CPG "in-band information or an appropriate pattern is now available" is interworked, a P-Early-Media header is sent* Ensure that the SUT, having received the ACM message called party status indicator "no indication", on receipt of a CPG message where the **event information parameter event indicator** is set to *"in-band information or an appropriate pattern is now available"*: • a 183 Session Progress is sent containing the P-Early-Media Header.	
SIP Parameter values:	183 Session Progress: P-Early-Media Header	
ISUP Parameter values:	ACM: Called party status "no indication" CPG; **event information parameter event indicator**: in-band-information or an appropriate pattern is now available	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) 183 Session Progress ← ← CPG (Inband info) 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP105005	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4** **ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	PICS 4/15 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*CPG Alerting received, 180 containing a P-Early-Media header is sent* Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the **event information parameter event indicator** is set to "Alerting" without BCI included: • sends an 180 Ringing response with P-Early Media	
SIP Parameter values:	180 Ringing PSTN XML **ProgressIndicator** PI_ID, P-Early-Media	
ISUP Parameter values:	CPG; **event information parameter event indicator**: Alerting	
Comments:	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← ← ACM (no indication) 180 Ringing ← ← CPG (Alerting) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP105006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4 ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*CPG Alerting contains an ATP with Progress Indicator* Ensure that SUT having received the ACM message, on receipt of a CPG message where the **event information parameter event indicator** is set to "Alerting" and the **ATP** containing the **progress indicator** PI_VALUE: • sends an 180 Ringing response with PSTN XML body containing containing PSTN XML ProgressIndicator set to PI_VALUE.	
SIP Parameter values:	180 Ringing PSTN XML **progress indicator** PI_ID	
ISUP Parameter values:	CPG; **event information parameter event indicator**: Alerting	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) 180 Ringing ← ← CPG (Alerting) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP105007	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4 ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*CPG Alerting containing a BCI and OBCI, mapping into PSTN XML instance* Ensure that the SUT having received the ACM message, on receipt of a CPG message where the **event information parameter event indicator** is set to " Alerting" the **ISUP indicator parameter** set to ISUP_ID, the **ISDN access indicator** set to **ISDN_ACCES_ID** and the OBCI in-band information set to OBCI_INBAND: • sends an 180 Ringing response with P-Early Media and PSTN XML body containing with the **progress indicator** information element set to PI_ID.	
SIP Parameter values:	180 Ringing PSTN XML **progress indicator** PI_ID	
ISUP Parameter values:	CPG; **event information parameter event indicator**: Alerting	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) 180 Ringing ← ← CPG (Alerting) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values for test purposes TP105007		
test purposes	**ISUP Parameter values:**	**PSTN XML progress descriptions:**
VA_01	**CPG** ISUP_ID: ISUP not used all the way	PI_ID: Call is not end-to-end ISDN (#1)
VA_02	**CPG** ISUP_ID: ISUP used all the way ISDN_ACCES_ID: non ISDN OBCI_INBAND: no	PI_ID: Destination address is non-ISDN (#2)
VA_03	**CPG** ISUP_ID: ISUP used all the way ISDN_ACCES_ID: non ISDN OBCI_INBAND: yes	PI_ID: Destination address is non-ISDN (#2) and In-band information or appropriate pattern now available (#8)
VA_04	**CPG** ISUP_ID: ISUP used all the way ISDN access indicator: ISDN	PI_ID: ("*Terminating access ISDN"(#7))0*.
VA_05	**CPG** ISUP_ID: ISUP used all the way ISDN access indicator: ISDN OBCI_INBAND: yes	PI_ID: In-band information or appropriate pattern now available (#8) and ("*Terminating access ISDN"(#7))*

TP105008	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4A** **ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*CPG Progress containing a BCI and OBCI, mapping into PSTN XML instance* Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the **event information parameter event indicator** is set to "Progress" and the **ISUP indicator parameter** set to ISUP_ID, the **ISDN access indicator** set to ISDN_ACCESS_ID and the OBCI in-band information set to OBCI_INBAND: • sends a 183 Session Progress with PSTN XML body containing the **progress indicator** set to PI_ID.	
SIP Parameter values:	183 Session Progress PSTN XML **progress indicator** PI_ID	
ISUP Parameter values:	CPG; **event information parameter event indicator**: Progress **ISUP indicator**: CPG_ISUP_ID **ISDN access indicator:** CPG_ISDN_ACCESS_ID	
Comments:	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← ← ACM (no indication) 183 Session Progress ← ← CPG (Progress) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values for test purposes TP105008		
Test purposes	**ISUP Parameter values:**	**ISDN Parameter values:**
VA_01	**CPG** CPG_ISUP_ID: ISUP not used all the way	PI_ID: Call is not end-to-end ISDN (#1)
VA_02	**CPG** CPG_ISUP_ID: ISUP used all the way CPG_ISDN_ACCES_ID: non ISDN	PI_ID: Destination address is non-ISDN (#2)
VA_03	**CPG** CPG_ISUP_ID: ISUP used all the way CPG_ISDN access indicator: ISDN	PI_ID: ("*Terminating access ISDN"(#7)*.

TP105009	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4A** **ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	PICS 4/15 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*CPG In-band information or an appropriate pattern is now available containing a BCI and OBCI, mapping into PSTN XML instance and P-Early-Media header* Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the **event information parameter event indicator** is set to "In-band information or an appropriate pattern is now available", the ISUP indicator set to CPG_ISUP_ID and the ISDN access indicator set to CPG_ISDN_ACCESS_ID and if the I-MGCF has received the P-Early-Media header in the INVITE request, then the I-MGCF shall: • send a 183 Session Progress with P-Early Media and **PSTN XML** body containing the **ProgressIndicator** set to PI_ID.	
SIP Parameter values:	183 Session Progress with P-Early Media and PSTN XML ProgressIndicator: PI_ID	
ISUP Parameter values:	CPG; **event information parameter event indicator**: In-band information or an appropriate pattern is now available **ISUP indicator**: CPG_ISUP_ID **ISDN access indicator**: CPG_ISDN_ACCES_ID	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) 183 Session Progress ← ← CPG (In-band info) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values for test purposes TP105009		
test purposes	**ISUP Parameter values:**	**ISDN Parameter values:**
VA_01	**CPG** CPG_ISUP_ID: ISUP used all the way CPG_ISDN_ACCES_ID: ISDN	PI_ID: In-band information or appropriate pattern now available (#8) and "*Terminating access ISDN*"(#7).
VA_02	**CPG** CPG_ISUP_ID: ISUP used all the way CPG_ISDN_ACCES_ID: non-ISDN	PI_ID: In-band information or appropriate pattern now available (#8) and Destination address is non-ISDN (#2)
VA_03	**CPG** CPG_ISUP_ID: ISUP not used all the way	PI_ID: In-band information or appropriate pattern now available (#8) and Call is not end-to-end ISDN (#1)

TP105010	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.1.4.1 ETS 300 356-1 [28] clause 2.1.4**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*Fallback procedure: CPG Alerting received, mapping of TMU and BC in the included ATP into 180* Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the event information parameter event indicator is set to " Alerting" and the Transmission medium used is included with the value TMU_VALUE and the Access Transport Parameter is set to ATP_VALUE: • sends an 180 Ringing message with the **PSTN XML BearerCapability** encoded ISDN_BC_VALUE.	
SIP Parameter values:	INVITE; **first Bearer Capability**: INVITE_BC1 **second Bearer Capability**: INVITE_BC2 180 Ringing; PSTN XML BearerCapability: ISDN_TMU_VALUE	
ISUP Parameter values:	CPG; event information parameter event indicator: Alerting TMU: TMU_VALUE ATP: BC ATP_VALUE	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) 180 Ringing ← ← CPG (Alerting) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP105011	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.1.4.1**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*Fallback procedure: CPG Progress received, mapping of TMU and BC in the included ATP into a 183* Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the **event information parameter event indicator** is set to "Progress" and the **Transmission medium requirement** is included with the value TMU_VALUE and the **Access Transport Parameter** is set to ATP_VALUE: • sends a 183 Session Progress message with the **PSTN XML BearerCapability** encoded ISDN_BC_VALUE.	
SIP Parameter values:	INVITE; **first BearerCapability**: INVITE_BC1 **second BearerCapability**: INVITE_BC2 180 Session Progress; **PSTN XML BearerCapability**: ISDN_BC_VALUE	
ISUP Parameter values:	CPG; **event information parameter event indicator**: Progress **TMU**: TMU_VALUE **ATP**: BC: ATP_VALUE	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) 183 Session Progress ← ← CPG (Progress) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values and selection criteria for test purposes TP105010 TP105011

Test purposes	**ACM Parameter values**	**180 Ringing Parameter values:**	**INVITE parameter value**
VA_01	TMU_VALUE: speech ATP_VALUE: no BC	ISDN_BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements
VA_02	TMU_VALUE: 3,1 kHz ATP_VALUE: no BC	ISDN_BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements
VA_03	TMU_VALUE: speech ATP_VALUE: BC = 3,1 kHz	ISDN_BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements
VA_04	TMU_VALUE: speech ATP_VALUE: BC = speech	ISDN_BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements
VA_05	TMU_VALUE: 3,1 kHz ATP_VALUE: BC = speech	ISDN_BC_VALUE: speech	INVITE_BC1: speech INVITE_BC2: unrestricted digital information with tones and announcements
VA_06	TMU_VALUE: 3,1 kHz audio ATP_VALUE: BC = 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio	INVITE_BC1: 3,1 kHz audio INVITE_BC2: unrestricted digital information with tones and announcements

TP105012	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.1.4.1**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*CPG Alerting received, sending of received PI #5 and HLC in an ATP into a PSTN XML PI and HLC in the 180* Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the **event information parameter event indicator** is set to " Alerting" and containing an Access Transport Parameter including a **High Layer Compatibility** (HLC) and containing the **progress indicator** #5: "interworking has occurred and has resulted in a telecommunication service change": • sends an 180 Ringing message with a **PSTN XML ProgressIndicator** with the progress indication "interworking has occurred and has resulted in a telecommunication service change" (#5) and with the **PSTN XML HighLayerCapability**.	
SIP Parameter values:	INVITE: **PSTN XML HighLayerCompatibility**: HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT) 180 Ringing; **PSTN XML ProgressIndicator**: interworking has occurred and has resulted in a telecommunication service change (#5) **PSTN XML HighLayerCapability**: HLC_VALUE2 (PIXIT)	
ISUP Parameter values:	CPG; **event information parameter event indicator**: Alerting **progress indicator**: interworking has occurred and has resulted in a telecommunication service change (#5) **HLC**: HLC_VALUE2 (PIXIT)	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) 180 Ringing ← ← CPG (Alerting) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

<table>
<tr><th>TP105013</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
ES 283 027 [1], clause 7.2.3.2.1.4A</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 4/18</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>CPG Progress received, sending of received PI #5 and HLC in an ATP in a PSTN XML PI and HLC in the 183</i>

Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the event information parameter event indicator is set to "Progress" and containing an Access Transport Parameter (ATP) including a High Layer Compatibility (HLC) and containing the progress indicator #5: "interworking has occurred and has resulted in a telecommunication service change":

• sends an 183 Session Progress message with a PSTN XML ProgressIndicator with the progress indication "interworking has occurred and has resulted in a telecommunication service change" (#5) and with the PSTN XML HighLayerCapability.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">INVITE: PSTN XML HighLayerCapability: HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT)
183 Session Progress; PSTN XML ProgressIndicator: interworking has occurred and has resulted in a telecommunication service change (#5)
PSTN XML HighLayerCapability: HLC_VALUE2 (PIXIT)</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5">CPG; event information parameter event indicator: Progress
ATP: progress indicator: interworking has occurred and has resulted in a telecommunication service change (#5)High Layer Capability: HLC_VALUE2 (PIXIT)</td></tr>
<tr><td rowspan="11">Comments:</td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>ACM (no indication)</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>183 Session Progress</td><td>←</td><td></td><td>←</td><td>CPG (Progress)</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP105014	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.1.4A**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*CPG Progress containing ATP PI, mapping into PSTN XML ProgressIndicator in the sent 183* Ensure that the SUT having received the ACM message, on receipt of a CPG message where the **event information parameter event indicator** is set to "Progress*",* the ATP contains the progress indicator set to PI-VALUE, the BCI **ISUP indicator parameter** set to ISUP used all the way and the BCI **ISDN access indicator** set to ISDN: • sends a 183 Session Progress message with the **PSTN XML ProgressIndicator** information element set to PI_ID.	
SIP Parameter values:	183 Session Progress; PSTN XML ProgressIndicator: PI_ID and "*Terminating access ISDN*"(#7)	
ISUP Parameter values:	ACM; **BCI ISUP indicator**: ISUP used all the way **BCI ISDN access indicator:** non ISDN CPG; **event information parameter event indicator**: Progress **BCI ISUP indicator**: ISUP used all the way **BCI ISDN access indicator:** ISDN ATP Progress Indicator value PI_ID	
Comments:	SIP / SUT / ISUP INVITE → → IAM ← ACM (no indication) 183 Session Progress ← ← CPG (Progress) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values and additional selection criteria for test purpose TP105014	
VA_01	PI_VALUE = Call is not end-to-end ISDN (#1)
VA_02	PI_VALUE = Destination address is non-ISDN (#2)

6.2.1.6 Receipt of the Answer message (ANM)

TP106001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5** **ETS 300 356-1 [28] clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*ANM received, a 200 OK INVITE is sent* Ensure that the SUT, having received the ACM message Called party status indicater set to "subscriber free", on receipt of an ANM message: • sends a 200 OK INVITE to the UAC.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM (free) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP106002	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5** **ETS 300 356-1 [28] clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ANM received, mapping of PI contained in the ATP into the 200 OK PSTN XML PI* Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing a **progress indicator** set to PI_VALUE in the ATP: • sends a 200 OK included the **PSTN XML ProgressIndicator** set to PI_VALUE.	
SIP Parameter values:	200 OK; **PSTN XML ProgressIndicator**: PI_VALUE (PIXIT)	
ISUP Parameter values:	ANM; **ATP progress indicator**: PI_VALUE (PIXIT)	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM (free) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP106003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5** **ETS 300 356-1 [28] clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*Fallback procedure: ANM received no BC in an ATP, mapping of TMU parameter into the PSTM XML PI sent in the 200 OK INVITE* Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing the Transmission Medium Used set to TMU_VALUE and the ATP without Bearer Capability (BC): • sends a 200 OK message with the **PSTN XML Bearer Capability** encoded ISDN_BC_VALUE.	
SIP Parameter values:	INVITE: **PSTN XML first Bearer Capability**: SETUP_BC1 **PSTN XML second Bearer Capability**: SETUP_BC2 200 OK; Bearer capability: ISDN_BC_VALUE	
ISUP Parameter values:	ANM; TMU: TMU_VALUE ATP: no BC	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM (free) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values for test purposes TP106003			
Test purposes	**ANM parameter values**	**200 OK parameter values**	**SETUP parameter values**
VA_01	TMU_VALUE: speech	ISDN_BC_VALUE: speech	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements
VA_02	TMU_VALUE: 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements

TP106004	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5** **ETS 300 356-1 [28] clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*Fallback procedure: ANM received BC in an ATP, mapping of TMU parameter and BC into the PSTM XML PI sent in the 200 OK INVITE* Ensure that the SUT , having received the ACM message, on receipt of an ANM message containing the **Transmission Medium Used** set to TMU_VALUE and the **ATP** set to ATP_VALUE and containing the **progress indicator** set to "interworking has occurred and has resulted in a telecommunication service change" (#5): • sends a 200 OK message with the **PSTN XML BearerCapability** encoded ISDN_BC_VALUE and the **PSTN XML ProgressIndicator** set to "interworking has occurred and has resulted in a telecommunication service change" (#5).	
SIP Parameter values:	INVITE: **PSTN XML first BearerCapability**: SETUP_BC1 **PSTN XML second Bearer Capability**: SETUP_BC2 200 OK; PSTN XML **BearerCapability**: ISDN_BC_VALUE **ProgressIndication**: interworking has occurred and has resulted in a telecommunication service change(#5)	
ISUP Parameter values:	ANM; **TMU**: TMU_VALUE **ATP**: ATP_VALUE **Progress indication**: interworking has occurred and has resulted in a telecommunication service change(#5)	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM (free) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values for test purposes TP106004			
Test purposes	**ACM parameter values**	**200 OK parameter values**	**SETUP parameter values**
VA_01	TMU_VALUE: speech ATP_VALUE: speech	ISDN_BC_VALUE: speech and **ProgressIndicator**: (#5)	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements
VA_02	TMU_VALUE: speech ATP_VALUE: 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio and **ProgressIndicator**: (#5)	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements
VA_03	TMU_VALUE: 3,1 kHz audio ATP_VALUE: speech	ISDN_BC_VALUE: speech and **ProgressIndicator**: (#5)	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements
VA_04	TMU_VALUE: 3,1 kHz audio ATP_VALUE: 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio and **ProgressIndicator**: (#5)	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements

TP106005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5 ETS 300 356-1 [28], clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*Fallback procedure: ANM received contains an ATP with BC unrestricted digital information with tones/announcement, mapping into the PSTN XML PI in the sent 200 OK INVITE* Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing the ATP including the **Bearer Capability** set to "unrestricted digital information with tones/announcement" and without **TMU** parameter: • sends a 200 OK message with the **PSTN XML Bearer Capability** set to "unrestricted digital information with tones/announcement".	
SIP Parameter values:	200 OK; PSTN XML **BearerCapabilty**: unrestricted digital information with tones/announcement	
ISUP Parameter values:	ANM; **ATP BC**: unrestricted digital information with tones/announcement **no TMU**	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM (free) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP106006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5 ETS 300 356-1 [28], clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ANM received contains an ATP parameter, mapping of HLC into the PSTN HLC in the 200 OK INVITE* Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing the HLC parameter in the **ATP** set to HLC_VALUE: • sends a 200 OK message **PSTN XML HighLayerCompatibility** information element set to HLC_VALUE.	
SIP Parameter values:	200 OK; **PSTN XML HighLayerCompatibility:** HLC_VALUE (PIXIT)	
ISUP Parameter values:	ANM; **ATP HLC**: HLC_VALUE	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM (free) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP106007	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5** **ETS 300 356-1 [28] clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*Fallback procedure: ANM received contains an ATP with HLC and PI #5, mapping into the PSTN XML PI in the sent 200 OK INVITE* Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing the HLC parameter in the **ATP** with an **HLC** set to HLC_VALUE and the **progress indicator** set to "interworking has occurred and has resulted in a telecommunication service change" (#5): - sends a 200 OK message with **the PSTN XML HighLayerCompatibility** information element set to HLC_VALUE and the **progress indicator** set to "interworking has occurred and has resulted in a telecommunication service change" (#5).	
SIP Parameter values:	INVITE: PSTN XML HighLayerCompatibility: HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT) 200 OK; PSTN XML HighLayerCompatibility**:** HLC_VALUE2 (PIXIT) PSTN XML P**rogressIndicator**: interworking has occurred and has resulted in a telecommunication service change (#5)	
ISUP Parameter values:	ANM; **ATP HLC**: HLC_VALUE2 **progress indicator**: interworking has occurred and has resulted in a telecommunication service change (#5)	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM (free) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP106008	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5** **ETS 300 356-1 [28] clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ANM received contains an ATP conveying the LLC, mapping into the PSTN XML LLC in the sent 200 OK INVITE* Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing the LLC parameter in the **ATP** set to LLC_VALUE: • sends a 200 OK message with a **PSTN XML LowLayerCompatibility** information element set to LLC_VALUE.	
SIP Parameter values:	200 OK INVITE: **PSTN XML LowLayerCompatibility:** LLC_VALUE (PIXIT)	
ISUP Parameter values:	ANM; **ATP LLC**: LLC_VALUE	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM (free) Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

6.2.1.7 Receipt of the Connect message (CON)

TP107001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CONNECT message (CON).	
SIP selection criteria:	PICS 4/1 AND PICS 4/4 AND PICS 4/5	
ISUP selection criteria:		
Test purpose:	*CON received, 200 OK INVITE is sent after Preconidions are met* **SDP offer was received** in the initial INVITE. Ensure that the SUT, on receipt of an CON message: sends a 200 OK INVITE to the UAC. The bearer path shall be connected in both directions when both of the following conditions are satisfied: • the I-MGCF determines (using the procedures defined in RFC 3312 [7]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "*notification not required*", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient preconditions have been met for the session to proceed.	
SIP Parameter values:	INVITE: Require: precondition SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv 183: Require: 100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv a=conf:qos remote sendrecv UPDATE: SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv 200 OK UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → → IAM 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE → → COT(successful) 200 OK UPDATE ← 200 OK INVITE ← ← CON ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP107002	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CONNECT message (CON).	
SIP selection criteria:	NOT PICS 4/1 AND PICS 4/4 AND PICS 4/5	
ISUP selection criteria:		
Test purpose:	*CON received, 200 OK INVITE is sent after Preconidions are met* **SDP offer was received** in the initial INVITE. Ensure that the SUT, on receipt of an CON message: sends a 200 OK INVITE to the UAC. The bearer path shall be connected in both directions when both of the following conditions are satisfied: • the I-MGCF determines (using the procedures defined in RFC 3312 [7]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "*notification not required*", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient preconditions have been met for the session to proceed.	
SIP Parameter values:	INVITE: Require: precondition SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv 183: Require: 100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv a=conf:qos remote sendrecv UPDATE: SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv 200 OK UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE → → IAM 200 OK UPDATE ← 200 OK INVITE ← ← CON ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP107003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CONNECT message (CON).	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CON received, 200 OK INVITE is sent* **SDP offer was received** in the initial INVITE. Ensure that the SUT, on receipt of an CON message: sends a 200 OK INVITE to the UAC. The bearer path shall be connected in both directions. In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "*notification not required*", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient preconditions have been met for the session to proceed.	
SIP Parameter values:	INVITE: SDP offer 200 OK INVITE: SDP answer	
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → → IAM 200 OK INVITE ← ← CON ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP107004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5 ETS 300 356-1 [28], clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*CON received contains a PI conveyed in an ATP, mapped into the PSTN XML PI sent in the 200 OK INVITE* Ensure that the SUT, on receipt of an CON message containing a **progress indicator** set to PI_VALUE in the ATP: • sends a 200 OK included the **PSTN XML ProgressIndicator** set to PI_VALUE.	
SIP Parameter values:	200 OK; **PSTN XML ProgressIndicator**: PI_VALUE (PIXIT)	
ISUP Parameter values:	ANM; **ATP progress indicator**: PI_VALUE (PIXIT)	
Comments:	SIP SUT ISUP INVITE → → IAM 200 OK INVITE ← ← CON ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP107005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5 ETS 300 356-1 [28], clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (CON).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*CON received contains the TMU parameter, mapping into the PSTN XML BC sent in the 200 OK INVITE* Ensure that the SUT, on receipt of an CON message containing the Transmission Medium Used set to TMU_VALUE and the ATP without Bearer Capability (BC): • sends a 200 OK message with the **PSTN XML BearerCapability** encoded ISDN_BC_VALUE.	
SIP Parameter values:	INVITE: **PSTN XML first Bearer Capability**: SETUP_BC1 **PSTN XML second Bearer Capability**: SETUP_BC2 200 OK; Bearer capability: ISDN_BC_VALUE	
ISUP Parameter values:	CON; TMU: TMU_VALUE ATP: no BC	
Comments:	SIP SUT ISUP INVITE → → IAM 200 OK INVITE ← ← CON ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Values for test purposes TP107005			
Test purposes	**CON parameter values**	**200 OK parameter values**	**SETUP parameter values**
VA_01	TMU_VALUE: speech	ISDN_BC_VALUE: speech	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements
VA_02	TMU_VALUE: 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements

TP107006	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5** **ETS 300 356-1 [28], clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (CON).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*CON received contains the TMU parameter and the ATP parameter conveying the PI #5, mapping into the PSTN XML PI sent in the 200 OK INVITE* Ensure that the SUT , on receipt of an CON message containing the **Transmission Medium Used** set to TMU_VALUE and the **ATP** set to ATP_VALUE and containing the **progress indicator** set to "interworking has occurred and has resulted in a telecommunication service change" (#5): • sends a 200 OK message with the **PSTN XML BearerCapability** encoded ISDN_BC_VALUE and the **PSTN XML ProgressIndicator** set to "interworking has occurred and has resulted in a telecommunication service change" (#5).	
SIP Parameter values:	INVITE: PSTN XML **first BearerCapability**: SETUP_BC1 **second Bearer Capability**: SETUP_BC2 200 OK; PSTN XML **BearerCapability**: ISDN_BC_VALUE **ProgressIndication**: interworking has occurred and has resulted in a telecommunication service change(#5)	
ISUP Parameter values:	CON; **TMU**: TMU_VALUE **ATP**: BC ATP_VALUE **Progress indication**: interworking has occurred and has resulted in a telecommunication service change(#5)	
Comments:	SIP SUT ISUP INVITE ➔ ➔ IAM 200 OK INVITE ← ← CON ACK ➔ Conversation BYE ➔ ➔ REL 200 OK BYE ← ← RLC	

Values for test purposes TP107006			
Test purposes	**ACM parameter values**	**200 OK parameter values**	**SETUP parameter values**
VA_01	TMU_VALUE: speech ATP_VALUE: BC speech	ISDN_BC_VALUE: speech and **ProgressIndicator**: (#5)	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements
VA_02	TMU_VALUE: speech ATP_VALUE: BC 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio and **ProgressIndicator**: (#5)	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements
VA_03	TMU_VALUE: 3,1 kHz audio ATP_VALUE: BC speech	ISDN_BC_VALUE: speech and **ProgressIndicator**: (#5)	SETUP_BC1: speech SETUP_BC2: unrestricted digital information with tones and announcements
VA_04	TMU_VALUE: 3,1 kHz audio ATP_VALUE: BC 3,1 kHz audio	ISDN_BC_VALUE: 3,1 kHz audio and **ProgressIndicator**: (#5)	SETUP_BC1: 3,1 kHz audio SETUP_BC2: unrestricted digital information with tones and announcements

TP107007	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5** **ETS 300 356-1 [28] clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (CON).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*CON received contains the ATP parameter conveying the BC unrestricted digital information with tones/announcement, mapping into the PSTN XML BC sent in the 200 OK INVITE* Ensure that the SUT, on receipt of an CON message containing the ATP including the **Bearer Capability** set to "unrestricted digital information with tones/announcement" and without **TMU** parameter: • sends a 200 OK message with the **PSTN XML Bearer Capability** set to "unrestricted digital information with tones/announcement".	
SIP Parameter values:	200 OK; **PSTN XML BearerCapabilty**: unrestricted digital information with tones/announcement	
ISUP Parameter values:	CON; **ATP BC**: unrestricted digital information with tones/announcement **no TMU**	
Comments:	INVITE → → IAM 200 OK INVITE ← ← CON ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP107008	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5** **ETS 300 356-1 [28], clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (CON).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*CON received contains the ATP parameter conveying the HLC, mapping into the PSTN XML HLC sent in the 200 OK INVITE* Ensure that the SUT, on receipt of an CON message containing the HLC parameter in the **ATP** set to HLC_VALUE: • sends a 200 OK message **PSTN XML HighLayerCompatibility** information element set to HLC_VALUE.	
SIP Parameter values:	200 OK; **PSTN XML HighLayerCompatibility:** HLC_VALUE (PIXIT)	
ISUP Parameter values:	CON; **ATP HLC**: HLC_VALUE	
Comments:	SIP SUT ISUP INVITE → → IAM 200 OK INVITE ← ← CON ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP107009	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5 ETS 300 356-1 [28], clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (CON).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*CON received contains the ATP parameter conveying the HLC and PI #5, mapping into the PSTN XML HLC sent in the 200 OK INVITE* Ensure that the SUT, on receipt of an CON message containing the HLC parameter in the **ATP** with an High Layer Compatibility set to HLC_VALUE and the **Progress.indicator** set to "interworking has occurred and has resulted in a telecommunication service change" (#5): • sends a 200 OK message with the **PSTN XML HighLayerCompatibility** information element set to HLC_VALUE and the **progress indicator** set to "interworking has occurred and has resulted in a telecommunication service change" (#5).	
SIP Parameter values:	INVITE: **PSTN XML HighLayerCompatibility**: HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT) 200 OK; **PSTN XML HighLayerCompatibility:** HLC_VALUE2 (PIXIT) **progress indicator**: interworking has occurred and has resulted in a telecommunication service change (#5)	
ISUP Parameter values:	CON; **ATP HLC**: HLC_VALUE2 **progress indicator**: interworking has occurred and has resulted in a telecommunication service change (#5)	
Comments:	SIP SUT ISUP INVITE → → IAM 200 OK INVITE ← ← CON ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP107010	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5 ETS 300 356-1 [28] clause 2.1.7**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (CON).	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*CON received contains an ATP conveying the LLC, mapping into the PSTN XML sent in the 200 OK INVITE* Ensure that the SUT, having received the ACM message, on receipt of an CON message containing the LLC parameter in the **ATP** set to LLC_VALUE: • sends a 200 OK message **wit a PSTN XML LowLayerCompatibility** information element set to LLC_VALUE.	
SIP Parameter values:	200 OK; **PSTN XML** LowLayerCompatibility**:** LLC_VALUE (PIXIT)	
ISUP Parameter values:	CON; **ATP LLC**: LLC_VALUE	
Comments:	SIP SUT ISUP INVITE → → IAM 200 OK INVITE ← ← CON ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

6.2.1.8 Receipt of the REL message

TP108001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*REL received after IAM was sent. Mapping into final response containing a Reason header* Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, on receipt of an ISUP REL, where the cause value defined as CV_ISUP: • the SUT immediately requests the disconnection of the internal bearer path. • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • The ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field.	
SIP Parameter values:	**cause value:** CV_SIP (PIXIT)	
ISUP Parameter values:	REL; **cause value:** CV_ISUP (PIXIT)	
Comments:	SIP SUT ISUP INVITE → → IAM SIP_FAILURE_VA ← ← REL ACK → → RLC	

Table 4

Values for test purposes TP108001		
←SIP Message SIP_FAILURE_VA CV_SIP		**← REL Cause Indicators parameter CV_ISUP**
VA_1	404 Not Found Cause Value No. 1	Cause Value No. 1 ("unallocated (unassigned) number")
VA_2	500 Server Internal Error Cause Value No. 2	Cause Value No. 2 ("no route to network")
VA_3	500 Server Internal Error Cause Value No. 3	Cause Value No. 3 ("no route to destination")
VA_4	500 Server Internal Error Cause Value No. 4	Cause Value No. 4 ("Send special information tone")
VA_5	404 Not Found Cause Value No. 5	Cause Value No. 5 ("Misdialled trunk prefix")
VA_6	500 Server Internal Error Cause Value No. 8	Cause Value No. 8 ("Preemption")
VA_7	500 Server Internal Error Cause Value No. 9	Cause Value No. 9 ("Preemption-circuit reserved for reuse")
VA_8	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_9	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("no user responding")
VA_10	480 Temporarily unavailable Cause Value No. 19	Cause Value No. 19 ("no answer from the user")
VA_11	480 Temporarily unavailable Cause Value No. 20	Cause Value No. 20 ("subscriber absent")
VA_12	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")
VA_13	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")
VA_14	480 Temporarily unavailable Cause Value No. 25	Cause Value No. 25 ("Exchange routing error")
VA_15	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")
VA_16	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete"))
VA_17	500 Server Internal Error	Cause Value No. 29 ("facility rejected")
VA_18	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_19	486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible) else 480 Temporarily unavailable Cause Value No. 34	Cause Value in the Class 010 (resource unavailable, Cause Value No. 34)
VA_20	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)
VA_21	500 Server Internal Error Cause Value No. 50	Cause Value No. 50 ("requested facility not subscribed")
VA_22	500 Server Internal Error Cause Value No. 55	Cause Value No. 55 ("incoming calls barred within CUG")
VA_23	500 Server Internal Error Cause Value No. 57	Cause Value No. 57 ("bearer capability not authorized")
VA_24	500 Server Internal Error Cause Value No. 58	Cause Value No. 58 ("bearer capability not presently")
VA_25	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_26	500 Server Internal Error Cause Value No. 65 - 79	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 - 79) (79 is class default)
VA_27	500 Server Internal Error Cause Value No. 87	Cause Value No. 87 ("user not member of CUG")

Values for test purposes TP108001		
←SIP Message SIP_FAILURE_VA CV_SIP		**← REL Cause Indicators parameter CV_ISUP**
VA_28	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")
VA_29	500 Server Internal Error Cause Value No. 90	Cause Value No. 90 ("Non-existent CUG")
VA_30	404 Not Found Cause Value No. 91	Cause Value No. 91 ("invalid transit network selection")
VA_31	500 Server Internal Error Cause Value No. 95	Cause Value No. 95 ("invalid message") (Class default)
VA_32	500 Server Internal Error Cause Value No. 97	Cause Value No. 97 ("Message type non-existent or not implemented")
VA_33	500 Server Internal Error Cause Value No. 99	Cause Value No. 99 ("information element/parameter non-existent or not implemented")
VA_34	480 Temporarily unavailable Cause Value No. 102	Cause Value No. 102 ("recovery on timer expiry")
VA_35	500 Server Internal Error Cause Value No. 103	Cause Value No. 103 ("Parameter non-existent or not implemented, pass on")
VA_36	500 Server Internal Error Cause Value No. 110	Cause Value No. 110 ("Message with unrecognized Parameter, discarded")
VA_37	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)
VA_38	480 Temporarily unavailable Cause Value No. 127	Cause Value No. 127 ("interworking unspecified") (Class default)

TP108002	**SIP reference: RFC 3261 [6]**			**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/			
SIP selection criteria:	PICS 4/15			
ISUP selection criteria:				
Test purpose:	*REL after ACM received, mapping in a final response* Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "**no indication**", on receipt of an ISUP REL, where the cause value defined as CV_ISUP: • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field.			
SIP Parameter values:	**cause value:** CV_SIP (PIXIT)			
ISUP Parameter values:	REL; **cause value:** CV_ISUP (PIXIT)			
Comments:	SIP INVITE SIP_FAILURE_VA ACK	 → ← →	SUT	ISUP → IAM ← ACM (no indication) ← REL → RLC

Table 5

Values for test purpose TP108002		
←SIP Message **SIP_FAILURE_VA** **CV_SIP**		**← REL** **Cause Indicators parameter** **CV_ISUP,**
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_2	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("No user responding")
VA_3	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")
VA_4	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")
VA_5	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")
VA_6	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete")
VA_7	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_8	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)
VA_9	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)

TP108003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*REL received in the early dialogue (ACM free) mapping in a final response* Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "**subscriber free**", having sent a 180 Ringing message on receipt of an ISUP REL, where the cause value defined as CV_ISUP: • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field.	
SIP Parameter values:	**Cause value:** CV_SIP (PIXIT)	
ISUP Parameter values:	REL; **Cause value:** CV_ISUP (PIXIT)	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM (free) SIP_FAILURE_VA ← ← REL ACK → → RLC	

TP108004	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:		
Test purpose:	*REL received in the early dialogue (CPG Alerting) mapping in a final response* Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "**no indication**", having received a CPG message where the **event information parameter event indicator** is set to "Alerting", a 180 Ringing message is sent, on receipt of an where the cause value defined as CV_ISUP: • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field.	
SIP Parameter values:	**Cause value:** CV_SIP (PIXIT)	
ISUP Parameter values:	REL; **cause value:** CV_ISUP (PIXIT)	
Comments:	SIP SUT ISUP INVITE → → IAM ← ACM (no indication) 180 Ringing ← ← CPG (Alerting) SIP_FAILURE_VA ← ← REL ACK → → RLC	

Table 6

Values for test purposes TP108003 and TP108004		
←SIP Message **SIP_FAILURE_VA** **CV_SIP**		**← REL** **Cause Indicators parameter** **CV_ISUP,**
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_2	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("No user responding")
VA_3	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")
VA_4	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")
VA_5	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")
VA_6	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete")
VA_7	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_8	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)
VA_9	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)

TP108005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*REL received in the confirmed state (ANM received)* Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message, having received an ANM', a 200 OK message is sent, on receipt of an ISUP REL where the cause value defined as CV_ISUP: • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send BYE message; • the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field in the BYE.	
SIP Parameter values:	**Cause value:** CV_SIP (PIXIT)	
ISUP Parameter values:	REL; **cause value:** CV_ISUP (PIXIT)	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP108006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*REL received in the confirmed state (CON received)* Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out a IAM message, having received a CON message, a 200 OK message is sent, on receipt of an where the cause value defined as CV_ISUP: • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send BYE message. • the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field.	
SIP Parameter values:	**Cause value:** CV_SIP (PIXIT)	
ISUP Parameter values:	REL; **cause value:** CV_ISUP (PIXIT)	
Comments:	SIP SUT ISUP INVITE → → IAM 200 OK INVITE ← ← CON ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

Table 7

Values for test purposes TP108005 and TP108006		
←SIP Message **SIP_FAILURE_VA** **CV_SIP**		**← REL** **Cause Indicators parameter** **CV_ISUP,**
VA_1	BYE Cause Value No. 16	Cause Value No. 16
VA_2	BYE Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_3	BYE Cause Value No. 38	Cause Value No. 38 ("Network out of order")
VA_4	BYE Cause Value No. 41	Cause Value No. 41 ("Temporary failure ")
VA_5	BYE Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)

6.2.1.9 Autonomous release at I-MGCF

TP109001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.10**
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-MGCF	
SIP selection criteria:		
ISUP selection criteria:	PICS 3/3 AND NOT PICS 3/4	
Test purpose:	*Overlap not supported, 484 is sent if insufficient digits received in the INVITE* Ensure that the SUT on receipt of insufficient digits received in an INVITE messages: • sends an 484 Address Incomplete message.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → 484 Address incomplete ← ACK →	

TP109002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.10**
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-MGCF	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*480 is sent if congestion in the SUT* Ensure that the SUT in congestion on receipt of INVITE message: • sends an 480 Temporarily Unavailable message.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → 480 Temporarily unavailable ← ACK →	

TP109003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.10**
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-MGCF	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*500 is sent to due the compatibility procedure for unknown parameters* Ensure that the call is released due to the BICC/ISUP compatibility procedure for unknown parameters: • sends 500 Server Internal Error.	
SIP Parameter values:		
ISUP Parameter values:	Unknown parameter in ACM: Parameter compatibility "Release call"	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM ← ACM (???) 500 Server internal error ← → REL ACK → ← RLC	

TP109004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.10**
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-MGCF	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Call setup is cleared after T7 expiry* Ensure that the call is released due to expiry of T7 within the BICC/ISUP procedures: • sends 484 Address Incomplete.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → → IAM Expiry of T7 484 Address incomplete ← → REL ACK → ← RLC	

TP109005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.10**
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-MGCF	
SIP selection criteria:		
ISUP selection criteria:	PICS 4/16	
Test purpose:	*Call setup is cleared after T9 expiry* Ensure that the call is released due expiry of T9 within the BICC/ISUP procedures: • sends 480 Temporarily Unavailable.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM Expiry of T9 480 Temporarily unavailable ← → REL ACK → ← RLC	

TP109006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.10**
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-MGCF	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*500 is sent to due the compatibility procedure for unknown messages* Ensure that the call is released due to the BICC/ISUP compatibility procedure for unknown messages: • sends 500 Server Internal Error.	
SIP Parameter values:		
ISUP Parameter values:	XXX: Unknown message: message compatibility "Release call"	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM ← XXX 500 Server internal error ← → REL ACK → ← RLC	

6.2.1.10 Receipt of the Release message BYE / CANCEL

TP110001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.6**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the BYE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*BYE with Reason header received, sending of REL* Ensure that the SUT on receipt of SIP BYE, the SUT shall send an ISUP REL to the ISUP side: • Ensure that the Reason header field with ITU-T Rec Q.850 [5] Cause Value **is included** in the BYE message is mapped to the ISUP Cause Value field in the ISUP REL message with the location "network beyond interworking point".	
SIP Parameter values:	Protocol-cause: CV_Reason Header (PIXIT)	
ISUP Parameter values:	REL: **cause value:** CV_ISUP (PIXIT)	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → BYE → → REL 200 OK BYE ← ← RLC	

TP110002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.6**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CANCEL message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CANCEL with Reason header received, sending of REL* Ensure that the SUT on receipt of SIP CANCEL, the I-MGCF shall send an ISUP REL to the ISUP side. • Ensure that the Reason header field with ITU-T Rec Q.850 [5] Cause Value **is included** in the CANCEL message is mapped to the ISUP Cause Value field in the ISUP REL message with the location "network beyond interworking point".	
SIP Parameter values:		
ISUP Parameter values:	REL: **cause value:** CV_ISUP (PIXIT) location: "network beyond interworking point"	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM CANCEL → → REL 200 OK CANCEL ← ← RLC 487 Request Terminated ← ACK →	

TP110003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.6**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the BYE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*BYE without Reason header received, sending of REL* Ensure that the SUT on receipt of SIP BYE without Reason header, the SUT shall send an ISUP REL to the ISUP side. Ensure that the the coding of the ISUP Cause Value is # 16 with the location "network beyond interworking point" if no reason header is containd in the SIP message.	
SIP Parameter values:		
ISUP Parameter values:	REL: **cause value:** #16	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → BYE → → REL 200 OK BYE ← ← RLC	

TP110004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.6**
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CANCEL message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CANCEL without Reason header received, sending of REL* Ensure that the SUT on receipt of SIP CANCEL without reason header, the SUT shall send an ISUP REL to the ISUP side. Ensure that the the coding of the ISUP Cause Value is # 31 with the location "network beyond interworking point" if no reason header is containd in the SIP message.	
SIP Parameter values:		
ISUP Parameter values:	REL: **cause value:** CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM CANCEL → → REL 200 OK CANCEL ← ← RLC 487 Request Terminated ← ACK →	

6.2.1.11 Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented

TP111001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.9**
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*RSC received in the confirmed state, a BYE is sent* Ensure that the SUT, when the communication is in the confirmed state, on receipt of a RSC message sends: • a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← RSC 200 OK BYE → → RLC	

TP111002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.9**
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*GRS received in the confirmed state, a BYE is sent* Ensure that the SUT, when the communication is in the confirmed state, on receipt of a GRS message sends: • a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← GRS 200 OK BYE → → GRA	

TP111003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.9**
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CGB "Hardware failure oriented" received in the confirmed state, a BYE is sent* Ensure that the SUT, when the communication is in the confirmed state, on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "**hardware failure oriented**", sends: • a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent.	
SIP Parameter values:		
ISUP Parameter values:	Circuit Group Supervision Message Type Indicator "hardware failure oriented"	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← CGB 200 OK BYE → → CGBA	

TP111004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.9**
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*RSC received after 200 OK INVITE was sent and ACK is not received* Ensure that the SUT, when an ANM was received, on receipt of a RSC message sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE. • the SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ← RSC ACK → → RLC BYE ← 200 OK BYE →	

TP111005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.9**
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*GRS received after 200 OK INVITE was sent and ACK is not received* Ensure that the SUT, when an ANM was received, on receipt of a GRS message sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE. • The SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ← GRS ACK → → GRA BYE ← 200 OK BYE →	

TP111006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.9**
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CGB "Hardware failure oriented" received after 200 OK INVITE was sent and ACK is not received* Ensure that the SUT, when an ANM was received, on receipt of a RSC message, with the Circuit Group Supervision Message Type Indicator coded as "**hardware failure oriented**", sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE. • The SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE.	
SIP Parameter values:		
ISUP Parameter values:	Circuit Group Supervision Message Type Indicator "hardware failure oriented"	
Comments:	SIP SUT ISUP INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ← CGB ACK → → CGBA BYE ← 200 OK BYE →	

<table>
<tr><th>TP111007</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.8.9</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>RSC in the early dialogue received, a 500 is sent</i>
Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a RSC message sends:
• a 500 Server Internal Error on the SIP side.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>Comments:</td><td colspan="2">SIP SUT ISUP
INVITE → → IAM
180 Ringing ← ← ACM
← RSC
480 Temporarily Unavailable ← → RLC
ACK →</td></tr>
</table>

<table>
<tr><th>TP111008</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.8.9</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>GRS in the early dialogue received, a 500 is sent</i>
Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends:
• a 500 Server Internal Error on the SIP side.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>Comments:</td><td colspan="2">SIP SUT ISUP
INVITE → → IAM
180 Ringing ← ← ACM
← GRS
480 Temporarily Unavailable ← → GRA
ACK →</td></tr>
</table>

TP111009	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.9**
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CGB "Hardware failure oriented" in the early dialogue received, a 500 is sent* Ensure that the SUT, when at least one backward ISUP message relating to the call has already been received on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented", sends • a 500 Server Internal Error on the SIP side.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM ← CGB 480 Temporarily Unavailable ← → CGBA ACK →	

TP111010	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8.9**
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*GRS for more than one connections received, a BYE is sent for each connection* Ensure that the SUT after receiving more than one INVITE sending an IAM message for each call association on receipt of a GRS message in the confirmed state, were the Range and Status Parameter value is bigger than *"1"*: • the SUT shall send a BYE requests for each call association.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE(1) → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → Conversation INVITE(2) → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → Conversation ← GRS BYE (1) ← → GRA 200 OK BYE → BYE (2) ← 200 OK BYE →	

<table>
<tr><th>TP111011</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.8.9</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>CGB "Hardware failure oriented" for more than one connections received, a BYE is sent for each connection</i>

Ensure that the SUT after receiving more than one INVITE sending an IAM message for each call association on receipt of a CGB message in the confirmed state, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" and were the Range and Status Parameter value is bigger than <i>"1"</i>:
the SUT shall send a BYE requests for each call association.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>SIP</th><th></th><th>SUT</th><th></th><th>ISUP</th></tr>
<tr><td>INVITE(1)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>INVITE(2)</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>CGB</td></tr>
<tr><td>BYE (1)</td><td>←</td><td></td><td>→</td><td>CGBA</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>BYE (2)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td></td><td></td></tr>
</table>
</td></tr>
</table>

6.2.1.12 Receipt of the Suspend message (SUS) network initiated

Void.

6.2.1.13 Receipt of the Resume message (RES) network initiated

Void.

6.2.2 Interworking from ISUP to SIP

6.2.2.1 Sending of the INVITE message

TP301001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.1.4**
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*IAM contains the complete Called party number and the sending complete indication, sending of INVITE* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** and the **sending complete** indication: • Sends the INVITE message.	
SIP Parameter values:		
ISUP Parameter values:	IAM; **Called party number**: with sending complete indication	
Comments:	**ISUP/BICC** SUT **SIP** IAM → → INVITE ACM ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301002	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.1.4**
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*IAM contains the maximum number of digits used in the national numbering plan, sending of INVITE* Ensure that the SUT in Idle state, on receipt of an IAM message containing the **maximum number of digits** used in the national numbering plan: • sends the INVITE message.	
SIP Parameter values:		
ISUP Parameter values:	IAM; **Called party number**: complete number	
Comments:	**ISUP/BICC** SUT **SIP** IAM → → INVITE ACM ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP301003</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.2.1.4</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP /Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>IAM contains a sufficient number of digits to route the call to the called party, sending of INVITE</i>
Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party:
• sends the INVITE message.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM; Called party number: complete number</td></tr>
<tr><td>Comments:</td><td colspan="2">ISUP/BICC SUT SIP
IAM → → INVITE
ACM ← ← 180 Ringing
Ringing tone
ANM ← ← 200 OK INVITE
→ ACK
Conversation
REL → → BYE
RLC ← ← 200 OK BYE</td></tr>
</table>

<table>
<tr><th>TP301004</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.2.1.4</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP /Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>IAM received, after timer Ti/w1 is expired , sending of INVITE</i>
Ensure that the SUT in Idle state, on receipt of an IAM message with the minimum number of digits required for routing the call have been received, by observing the timer Ti/w1 which has expired
• sends the INVITE message.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>Comments:</td><td colspan="2">ISUP/BICC SUT SIP
IAM →
$T_{I/W1}$ expiry
→ INVITE
ACM ← ← 180 Ringing
Ringing tone
ANM ← ← 200 OK INVITE
→ ACK
Conversation
REL → → BYE
RLC ← ← 200 OK BYE</td></tr>
</table>

<table>
<tr><th>TP301005</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.2.1.2</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 1/3</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>IAM received continuity check indicator is set to "continuity check not required", the INVITE is sent immediately</i>
Ensure that the SUT in Idle state, on receipt of an IAM message with the complete called party number containing the Continuity Check indicator in the Nature of Connection Indicators parameter is set to indicate "continuity check not required":
• sends a INVITE message.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM: Nature of Connection Indicators parameter is set to indicate "continuity check not required"</td></tr>
<tr><td>Comments:</td><td colspan="2"><table>
<tr><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ACM</td><td></td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table></td></tr>
</table>

<table>
<tr><th>TP301006</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.2.1.2</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">NOT PICS 4/11</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 4/5 AND PICS 1/3</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>IAM received continuity check indicator is set to " continuity check required on this circuit", the INVITE is sent after COT "successful" is received</i>
Ensure that the SUT in Idle state, on receipt of an IAM message with the complete called party number containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "<i>continuity check required on this circuit</i>":
• Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter <i>"continuity check successful"</i>.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM: Nature of Connection Indicators parameter which is set to "<i>continuity check required on this circuit</i>"
COT: Continuity Indicators parameter <i>"continuity check successful"</i></td></tr>
<tr><td>Comments:</td><td colspan="2"><table>
<tr><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>COT(successful)</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table></td></tr>
</table>

TP301007	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.1.2**
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND NOT PICS 4/11	
ISUP selection criteria:	PICS 1/3	
Test purpose:	*IAM received continuity check indicator is set to* ***"continuity check performed on previous circuit"****, the INVITE is sent after COT "successful" is received* Ensure that the SUT in Idle state, on receipt of an IAM message with the complete **called party number** containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to **"*continuity check performed on previous circuit*"**: • Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "***continuity check successful***"**.**	
SIP Parameter values:		
ISUP Parameter values:	IAM Nature of Connection Indicators parameter which is set to **"*continuity check performed on previous circuit*"** COT: Continuity Indicators parameter ***"continuity check successful"***	
Comments:	**ISUP/BICC** SUT **SIP** IAM → COT(successful) → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301008	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.1.2, 7.2.3.2.3**
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/11	
ISUP selection criteria:	PICS 1/3 AND PICS 4/2	
Test purpose:	*IAM continuity check required received, precondition request in the INVITE* Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate **"*continuity check required on this circuit*"**: • sends an INVITE message with precondition using the SDP offer in the INVITE. The SDP offer or answer carrying the confirmation of a precondition being met is sent when the Continuity message with the Continuity Indicators parameter set to **"*continuity check successful*"**was received and the requested preconditions are met in the SIP network.	
SIP Parameter values:	INVITE: Require: precondition SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv 183: Require: 100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv a=conf:qos remote sendrecv UPDATE: SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv 200 OK UPDATE SDP a=curr:qos local sendrecv a=curr:qos remote sendrecv a=des:qos mandatory local sendrecv a=des:qos mandatory remote sendrecv	
ISUP Parameter values:	IAM: Nature of Connection Indicators parameter which is set to **"*continuity check required on this circuit*"** COT: Continuity Indicators parameter ***"continuity check successful"***	
Comments:	**ISUP/BICC** SUT **SIP** IAM → → INVITE ← 183 Session Progress → PRACK ← 200 OK PRACK COT → → UPDATE ← 200 OK UPDATE ACM ← ← 180 Ringing → PRACK ← 200 OK PRACK Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP301009</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.2.1.2, 7.2.3.2.3</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 4/5 AND PICS 4/11</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 1/3 AND PICS 4/2</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>IAM continuity check performed on a previous circuit received, precondition request in the INVITE</i>

Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate "<i>continuity check performed on previous circuit</i>":

• sends an INVITE message with precondition using the SDP offer in the INVITE. The SDP offer or answer carrying the confirmation of a precondition being met is sent when the Continuity message with the Continuity Indicators parameter set to "<i>continuity check successful</i>" was received and the requested preconditions are met in the SIP network.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INVITE: Require: precondition
SDP a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos none remote sendrecv

183: Require: 100rel
SDP a=curr:qos local none
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv
a=conf:qos remote sendrecv

UPDATE:
SDP a=curr:qos local sendrecv
a=curr:qos remote none
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv

200 OK UPDATE
SDP a=curr:qos local sendrecv
a=curr:qos remote sendrecv
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendrecv</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM Nature of Connection Indicators parameter which is set to "<i>continuity check performed on previous circuit</i>"
COT: Continuity Indicators parameter <i>"continuity check successful"</i></td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>ISUP/BICC</th><th></th><th>SUT</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>
</td></tr>
</table>

TP301010	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.1.3**
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/20	
ISUP selection criteria:		
Test purpose:	*Support of Information Request message (INR)* Ensure that if no calling party number is received in the incoming IAM message, the O-MGCF sends an INR message to request the calling party number and not sends the INVITE request until receiving an INF message with calling party number. If no calling party number is received in the INF message, O-MGCF may reject or continue the call based on local configuration.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** SUT **SIP** IAM → INR ← INF → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><td>TP301011</td><td>SIP reference: RFC 3261 [6]</td><td>ISUP reference:
ES 283 027 [1], clause 7.2.3.2.1a</td></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/Basic call/Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Sending of INVITE without determining the end of address signalling</i>

Ensure that if the O-MGCF sends an INVITE request before the end of address signalling is determined, the O-MGCF shall:
- start timer Ti/w2; and
- be prepared to process SAM
- be prepared to handle incoming SIP 404 or 484 error.

On receipt of a SAM from the ISUP side, the O-MGCF shall:
stop timer Ti/w3 (if it is running);
send an INVITE request complying to the following:
- The INVITE request shall include all digits received so far for this call in the Request-URI. restart Ti/w2.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>404/484</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>404/484</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>
</td></tr>
</table>

TP301012	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.2.2**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	Based on table 8	
ISUP selection criteria:		
Test purpose:	*Mapping of TMR into the SDP in the sent INVITE* Ensure that the SUT in the Idle state on receipt of a IAM message, with the **Transmission Medium Requirement** (TMR) parameter set to TMR_VALUE: • sends an INVITE message containing the media description defined with the "a=" "b=" and "m=" lines set to a_b_m_LINE_VALUE.	
SIP Parameter values:	INVITE: a_b_m_LINE_VALUE	
ISUP Parameter values:	IAM: TMR: ISUP_TMR	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301013	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.2.2**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	Based on table 9	
ISUP selection criteria:		
Test purpose:	*Mapping of USI into the SDP in the sent INVITE* Ensure that the SUT in the Idle state on receipt of an IAM message, with the **user information parameter** set to USI_VALUE: • sends an INVITE message, with the media description defined with the "a = " "b=" and "m=" lines set to a_b_m_LINE_VALUE.	
SIP Parameter values:	INVITE: a_b_m_LINE_VALUE	
ISUP Parameter values:	IAM: USI: ISUP_USI	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

P301014	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.2**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 2/8	
ISUP selection criteria:		
Test purpose:	*SDP offer using the AMR codec* Ensure that the SUT in the Idle state on receipt of an IAM message indicating a speech call, with the **user information parameter** set to USI_VALUE: • sends an INVITE message, with the AMR codec.	
SIP Parameter values:	Offer: m=audio RTP/AVP dynimac PT a = rtpmap dynimac PT AMR Answer: m=audio RTP/AVP dynimac PT a = rtpmap dynimac PT AMR	
ISUP Parameter values:	IAM: USI= USI_VALUE (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301015	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.2**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	NOT PICS 2/8	
ISUP selection criteria:		
Test purpose:	*No AMR codec in the SDP, when no equipment implements the AMR codec* Ensure that the SUT in the Idle state on receipt of an IAM message indicating a speech call, with the user information parameter set to USI_VALUE and the IMS network serves that no user equipment implements the AMR codec, then the AMR codec shall be excluded from the SDP offer.	
SIP Parameter values:	INVITE: SDP no AMR codec	
ISUP Parameter values:	IAM: USI= USI_VALUE (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Table 8

Values for test purposes TP301012						
	ISUP	**SDP - a_b_m_LINE_VALUE**				
	TMR parameter	**m= line**			**b= line**	**a= line**
	TMR codes	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>: <bandwidth-value>**	**rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>**
VA_01	"speech"	Audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
VA_02	"speech"	Audio	RTP/AVP	Dynamic PT (and possibly a second Dynamic PT)	AS:64	rtpmap:<dynamic-PT> PCMU/8000 (and possibly rtpmap:<dynamic-PT> PCMA/8000)
VA_03	"speech"	Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_04	"speech"	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> PCMA/8000
VA_05	"3,1 KHz audio"	Audio	RTP/AVP	0 and/or 8	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000
VA_06	"3,1 KHz audio"	Audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
VA_07	"3,1 KHz audio"	Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_08	"64 kbit/s unrestricted"	Audio	RTP/AVP	9	AS:64	rtpmap:9 G722/8000
VA_9	"64 kbit/s unrestricted"	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> CLEARMODE/8000

Table 9

Values for test purposes TP301012, TP301013									
VA	**ISUP**				**SDP - a_b_m_LINE_VALUE**				
		USI parameter		**HLC IE in ATP**	**m= line**			**b= line**	**a= line**
	TMR	**Information Transport Capability**	**User Information Layer 1 Protocol Indicator**	**High Layer Characteristics Identification**	**<media>**	**<transport>**	**<fmt-list>**	**<modifier>:<bandwidth-value>**	**rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>**
VA_01	*"speech"*	*"Speech"*	*"G.711 µ-law"*	*Ignore*	*audio*	*RTP/AVP*	*0 (and possibly 8) (NOTE 1)*	*AS:64*	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000) (see note 1)
VA_02	*"speech"*	*"Speech"*	*"G.711 µ-law"*	*Ignore*	*audio*	*RTP/AVP*	*Dynamic PT (and possibly a second Dynamic PT) (NOTE 1)*	*AS:64*	rtpmap:<dynamic-PT> PCMU/8000 (and possibly rtpmap:<dynamic-PT> PCMA/8000) (see note 1)
VA_03	*"speech"*	*"Speech"*	*"G.711 A-law"*	Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_04	*"speech"*	*"Speech"*	*"G.711 A-law"*	Ignore	audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> PCMA/8000
VA_05	*"3,1 kHz audio"*	USI Absent		Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_06	*"3,1 kHz audio"*	*"3,1 kHz audio"*	*"G.711 µ-law"*	(NOTE 3)	audio	RTP/AVP	0 (and possibly 8) (NOTE 1)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000) (see note 1)
VA_07	*"3,1 kHz audio"*	*"3,1 kHz audio"*	*"G.711 A-law"*	(NOTE 3)	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_08	*"3,1 kHz audio"*	*"3,1 kHz audio"*		*"Facsimile Group 2/3"*	image	udptl	t38	AS:64	Based on ITU-T Recommendation T.38 [31].
VA_09	*"3,1 kHz audio"*	*"3,1 kHz audio"*		*"Facsimile Group 2/3"*	image	tcptl	t38	AS:64	Based on ITU-T Recommendation T.38 [31]
VA_10	*"64 kbit/s unrestricted"*	*"Unrestricted digital inf. W/tone/ann."*	N/A	Ignore	audio	RTP/AVP	9	AS:64	rtpmap:9 G722/8000
VA_11	*"64 kbit/s unrestricted"*	*"Unrestricted digital information"*	N/A	Ignore	audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> CLEARMODE/8000 (see note 2)

NOTE 1 Both PCMA and PCMU could be required.

NOTE 2 CLEARMODE is specified in RFC 4040 [19].

NOTE 3 HLC is normally absent in this case. It is possible for HLC to be present with the value "Telephony", although 6.3.1/Q.939 indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.

<table>
<tr><th>TP301016</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.2.2.2</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/Basic call/ Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 2/8</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>The SUT defines the RS and RR bandwidth modifiers when AMR codec is used</i>
Ensure that the if the O-MGCF determines that a speech call is incoming, the O-MGCF shall include the AMR codec and provides SDP RR and RS bandwidth modifiers specified in RFC 3556 [17] to disable RTCP.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INVITE: SDP b=RS:<bandwidth-value>
b=RR:<bandwidth-value></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>Comments:</td><td colspan="2">ISUP/BICC SUT SIP
IAM → → INVITE
ACM ← ← 180 Ringing
Ringing tone
ANM ← ← 200 OK INVITE
→ ACK
Conversation
REL → → BYE
RLC ← ← 200 OK BYE</td></tr>
</table>

<table>
<tr><th>TP301017</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.2.2.1</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/Basic call/ Sending of the INVITE message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Mapping of Called party number into the To header user=phone is included</i>
Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM:
• to the addr-spec component of the To header field which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INVITE: To: sip: ….; user=phone</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>Comments:</td><td colspan="2">ISUP/BICC SUT SIP
IAM → → INVITE
ACM ← ← 180 Ringing
Ringing tone
ANM ← ← 200 OK INVITE
→ ACK
Conversation
REL → → BYE
RLC ← ← 200 OK BYE</td></tr>
</table>

TP301018	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.1**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of Called party number into the To header received digits in the addr-spc component* Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM and the and the followed SAM: • to the addr-spec component of the **To header field**.	
SIP Parameter values:	INVITE: To:	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301019	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.1**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of Called party number into the To header as a SIP URI* Ensure that the SUT is mapping in the Called Party Number parameter contained in the Called Party address information of the IAM and followed SAM: • to the addr-spec component of the **To header field** which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.	
SIP Parameter values:	INVITE: To: sip: ….; user=phone	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 404/484 → ACK SAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301020	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.4**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:	PICS 4/5	
Test purpose:	*Mapping of HOP counter into the Max-Forwards header* Ensure that if the Hop Counter procedure is supported in the CS network, the O-MGCF shall use the Hop Counter parameter to derive the Max-Forwards SIP header. Due to the different default values (that are based on network demands/provisions) of the SIP Max-Forwards header and the Hop Counter, an adaptation mechanism shall be used to adopt the Hop Counter to the Max Forwards at the O-MGCF. Max-Forwards for a given message should be monotone decreasing with each successive visit to a SIP entity, regardless of intervening interworking, and similarly for Hop Counter.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:		

ISUP/BICC		SUT		SIP
IAM	→		→	INVITE
ACM	←		←	180 Ringing
		Ringing tone		
ANM	←		←	200 OK INVITE
			→	ACK
		Conversation		
REL	→		→	BYE
RLC	←		←	200 OK BYE

Table 10: Hop counter-Max forwards

Hop Counter	= X	Max-Forwards	= Y = Integer part of (X * Factor)
NOTE: The Mapping of value X to Y should be done with the used (implemented) adaptation mechanism.			

TP301021	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.1**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:	PICS 1/6	
Test purpose:	*Mapping of a "international number" into the To header and Request URI* Ensure that the called party number parameter of the IAM message is used to derive Request URI and To header of the INVITE Request. If the Request URI is a tel URI with "user=phone" it shall contain an International public telecommunication number prefixed by a "+" sign (e.g. tel:+4911231234567). If the Request URI is a sip URI with "user=phone" it shall contain an International public telecommunication number prefixed by a "+" sign and a host portion (e.g. sip:+4911231234567@host). Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, **Nature of address = "International number"** of the IAM: • to the format of the To header field and Request URI inserting "+" before the Addess signal.	
SIP Parameter values:	INVITE: To: …, Request URI	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301022	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.1**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 1/6	
Test purpose:	*Mapping of Called party number "national (significant) number" into the To header and Request URI user=phone is included* Ensure that the called party number parameter of the IAM message is used to derive Request URI of the INVITE Request. The Request URI is a tel URI with "user=phone" and shall contain an International public telecommunication number prefixed by a "+" sign (e.g. tel:+4911231234567). If the Request URI is a sip URI with "user=phone" it shall contain an International public telecommunication number prefixed by a "+" sign and a host portion (e.g. sip:+4911231234567@host). Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, **Nature of address = "National (significant) number"** of the IAM: • to the format of the To header field and Request URI inserting "+" CC before the Addess signal; • the format of the To header field is "+CC+NDC+SN".	
SIP Parameter values:	INVITE: To: …	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301023	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.7
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of USI parameter into PSTN XML BearerCapability* Ensure that the SUT in Idle state, on receipt of an IAM message **User Service Information** (USI) set to USI_VALUE: • sends the INVITE message with the PSTN XML **Bearer Capability** (BC) set to USI_VALUE.	
SIP Parameter values:	INVITE; PSTN XML **BearerCapability** (BC): USI_VALUE (PIXIT)	
ISUP Parameter values:	IAM; **USI:** USI_VALUE (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301024	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.7
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of USI and USI prime parameter into PSTN XML BearerCapability* Ensure that the SUT in Idle state, on receipt of an IAM message **USI** set to "speech" and the **USI Prime** set to "unrestricted digital information with tones and announcements": • sends the INVITE message with the **first PSTN XML BearerCapability** set to "speech" (the USI value) and the **second PSTN XML BearerCapability** set to "unrestricted digital information with tones and announcements" (the USI prime value).	
SIP Parameter values:	INVITE; **first PSTN XML Bearer Capability**: speech **second PSTN XML Bearer Capability**: unrestricted digital information with tones and announcements	
ISUP Parameter values:	IAM; **USI:** speech **USI Prime**: unrestricted digital information with tones and announcement	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

P301025	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.2.8**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of FCI "ISUP not used all the way" into PSTN XML ProgressIndicator #1* Ensure that the SUT in Idle state, on receipt of an IAM message containing the **ISUP indicator** set to "ISUP not used all the way": • sends INVITE message with the PSTN XML ProgressIndicator set to "call is not end-to-end ISDN: further call progress information is available in-band (#1)".	
SIP Parameter values:	INVITE; **PSTN XML ProgressIndicator**: call is not end-to-end ISDN: further call progress information is available in-band (#1)	
ISUP Parameter values:	IAM; **ISUP indicator**: ISUP not used all the way	
Comments:	**ISUP/BICC** SUT **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301026	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.2.8**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of FCI "Originating access non ISDN" into PSTN XML ProgressIndicator #3* Ensure that the SUT in Idle state, on receipt of an IAM message containing the **ISUP indicator** set to "ISUP used all the way" and the **ISDN access indicator** set to "originating access non-ISDN": • sends the INVITE message with the **PSTN XML ProgressIndicator** se tot "Originating access is non ISDN (#3)".	
SIP Parameter values:	INVITE: **PSTN XML ProgressIndicator**: Originating access is non ISDN (#3)	
ISUP Parameter values:	IAM; **ISUP indicator**: ISUP used all the way **ISDN access indicator**: originating access non-ISDN	
Comments:	**ISUP/BICC** SUT **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301027	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.8**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of FCI "ISUP used all the way" and "Originating access non ISDN" into PSTN XML ProgressIndicator #3* Ensure that the SUT in Idle state, on receipt of an IAM message containing the **ISUP indicator** set to "ISUP used all the way" and the **ISDN access indicator** set to "originating access non-ISDN", • sends the INVITE message with the **PSTN XML ProgressIndicatorPSTN XML ProgressIndicator** set to "Originating access is non ISDN (#3)".	
SIP Parameter values:	INVITE; **PSTN XML ProgressIndicatorPSTN XML ProgressIndicator**: Originating access id non ISDN (#3)	
ISUP Parameter values:	IAM; **ISUP indicator**: ISUP used all the way **ISDN access indicator**: originating access non-ISDN	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301028	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.8**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of FCI "ISUP used all the way" and "Originating access ISDN" into PSTN XML ProgressIndicator #6* Ensure that the SUT in Idle state, on receipt of an IAM message containing the **ISUP indicator** set to "ISUP used all the way" and the **ISDN access indicator** set to "originating access ISDN" • sends the INVITE message with the **PSTN XML ProgressIndicator** set to "originating access ISDN" (#6).	
SIP Parameter values:	INVITE; **PSTN XML ProgressIndicator**: "originating access ISDN" (#6)	
ISUP Parameter values:	IAM; **ISUP indicator**: ISUP used all the way **ISDN access indicator**: originating access ISDN	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301029	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.8**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of FCI "ISUP used all the way" and "Originating access ISDN" and ATP contains a Progress Indicator into PSTN XML ProgressIndicator #6* Ensure that the SUT in Idle state, on receipt of an IAM message containing the **ISUP indicator** set to "ISUP used all the way" and the **ISDN access indicator** set to "originating access ISDN" and an Access Transport Parameter (ATP) containing progress indicator set to PI_VALUE, • sends the INVITE message with the **PSTN XML ProgressIndicator** set to "originating access ISDN" (#6) and PI_VALUE.	
SIP Parameter values:	INVITE; **PSTN XML ProgressIndicator**: "originating access ISDN" (#6) and **PSTN XML ProgressIndicator**: PI_VALUE (PIXIT)	
ISUP Parameter values:	IAM; **ISUP indicator**: ISUP used all the way **ISDN access indicator**: originating access ISDN **ATP progress indicator**: PI_VALUE (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301030	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.8**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of ATP contains a LLC into a PSTN XML LLC in the sent INVITE* Ensure that the SUT in Idle state, on receipt of an IAM message containing the Access Transport Parameter (ATP) containing the **Low Layer Compatibility** (LLC) set to LLC_VALUE, • sends the INVITE message with the **PSTN XML LowLayerCompatibility** set to LLC_VALUE.	
SIP Parameter values:	the **PSTN XML LowLayerCompatibility**: LLC_VALUE (PIXIT)	
ISUP Parameter values:	IAM; **ATP LLC**: LLC_VALUE (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301031	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.2.8**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of ATP contains a HLC into a PSTN XML HLC in the sent INVITE* Ensure that the SUT in Idle state, on receipt of an IAM message containing the Access Transport Parameter (ATP) containing the **High Layer Compatibility** (HLC) set to HLC_VALUE, • sends the INVITE message with the **PSTN XML HighLayerCompatibility** set to HLC_VALUE.	
SIP Parameter values:	INVITE: **PSTN XML HighLayerCompatibility**: HLC_VALUE (PIXIT)	
ISUP Parameter values:	IAM; **ATP HLC**: HLC_VALUE (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301032	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.2.8**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of User Teleservice Information parameter into a PSTN XML HLC in the sent INVITE* Ensure that the SUT in Idle state, on receipt of an IAM message containing the User Teleservice containing the **High Layer Compatibility** (HLC) set to HLC_VALUE, • sends the INVITE message with the **PSTN XML HighLayerCompatibility** set to HLC_VALUE.	
SIP Parameter values:	INVITE: **PSTN XML HighLayerCompatibility**: HLC_VALUE (PIXIT)	
ISUP Parameter values:	IAM; **User Teleservice Information**: HLC_VALUE (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301033	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.8**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*Mapping of two HLC parameter contained in an ATP parameter into two PSTN XML HighLayerCompatibility elements* Ensure that the SUT in Idle state, on receipt of an IAM message containing the Access Transport Parameter (ATP) containing two **High Layer Compatibility** (HLC) set to respectively HLC_VALUE1 and HLC_VALUE2, • sends the INVITE message with two **PSTN XML HighLayerCompatibility** in the same order HLC_VALUE1 and HLC_VALUE2.	
SIP Parameter values:	INVITE; **first PSTN XML HighLayerCompatibility**: HLC_VALUE1 (PIXIT) **second PSTN XML HighLayerCompatibility**: HLC_VALUE2 (PIXIT)	
ISUP Parameter values:	IAM; **ATP first HLC**: HLC_VALUE1 (PIXIT) **ATP second HLC**: HLC_VALUE2 (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP301034	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.2.8**
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of calling party category into cpc parameter in the P-Aserted-Identity* Ensure that the SUT map the calling party category ISUP_CPC into the cpc SIP_CPC parameter in the P-Asserted-Identity and Accept-Contact header parameter "language" SIP_LANG.	
SIP Parameter values:	INVITE; P-Asserted-Identity, Accept-Contact	
ISUP Parameter values:	IAM; Calling party category	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Values for test purposes TP301034		
ISUP_CPC	**SIP Parameters**	
received calling party's category	**SIP_CPC in P-Asserted-Identity**	**SIP_LANG Accept-Contact 'language'**
operator, language French	operator	French
operator, language English	operator	English
operator, language German	operator	German
operator, language Russian	operator	Russian
operator, language Spanish	operator	Spanish
ordinary calling subscriber	ordinary	
test call	test	
payphone	payphone	
mobile terminal located in the home PLMN	cellular	
mobile terminal located in a visited PLMN	cellular roaming	
IEPS call marking for preferential call set up	ieps	

6.2.2.2 Receipt of the SAM message after INVITE has been send

TP302001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.1.4**
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after INVITE has been sent	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	PICS 3/5 AND NOT PICS 3/8	
Test purpose:	*Overlap procedure not supported, SAM is ingnored* Ensure if the SUT is supporting en bloc addressing towards the SIP network, subsequent SAMs received after the SUT has sent the INVITE are ignored.	
SIP Parameter values:		
ISUP Parameter values:	SAM; **subsequent number** (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM ➔ ➔ INVITE SAM ➔ ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE ➔ ACK Conversation ➔ ➔ BYE RLC ← ← 200 OK BYE	

TP302002	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clauses 7.2.3.2.1a and 7.2.3.2.1.4**
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2	
ISUP selection criteria:	PICS 3/8	
Test purpose:	*Overlap procedure supported by determining the end of address signalling. sending complete indication received* Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to indicate "**continuity check not required**" on receipt of a SAM containing the complete **called party number** and the **sending complete** indication and the SUT On receipt of a SAM from the ISUP the SUT shall: Stop timer TOIW3 (if it is running). TOIW2 shall be restarted and the SUT shall invoke the following procedures: a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-MGCF may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 404/484 → ACK SAM → → INVITE ← 404/484 → ACK SAM → → INVITE ← 404/484 → ACK SAM(F) → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP302003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clauses 7.2.3.2.1a and 7.2.3.2.1.4**
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2	
ISUP selection criteria:	PICS 3/8	
Test purpose:	*Overlap procedure supported by determining the end of address signalling. Maximum number of digits used in the national numbering plan are reached* Ensure that the SUT in Idle state, on receipt of an IAM message containing Continuity Check indicator in the Nature of Connection Indicators parameter which is set to indicate "**continuity check not required**" on receipt of a SAM and the **maximum number of digits used in the national numbering plan** are reached, the SUT sends an INVITE message containing all digits received in the IAM and the SAM(s). Stop timer TOIW3 (if it is running). TOIW2 shall be restarted and the SUT shall invoke the following procedures: a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-MGCF may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 404/484 → ACK SAM → → INVITE ← 404/484 → ACK SAM → → INVITE ← 404/484 → ACK SAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP302004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clauses 7.2.3.2.1a and 7.2.3.2.1.4**
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2	
ISUP selection criteria:	PICS 3/8	
Test purpose:	*Overlap procedure supported by determining the end of address signalling. Sufficient number of digits has been received to route the call received* Ensure that the SUT in Idle state, on receipt of an IAM message containing Continuity Check indicator in the Nature of Connection Indicators parameter which is set to indicate "**continuity check not required**" on receipt of a SAM and the **sufficient number of digits has been received to route the call** to the called party, the SUT sends an INVITE message containing all digits received in the IAM and the SAM(s). Stop timer TOIW3 (if it is running). TOIW2 shall be restarted and the SUT shall invoke the following procedures: a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-MGCF may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 404/484 → ACK SAM → → INVITE ← 404/484 → ACK SAM → → INVITE ← 404/484 → ACK SAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP302005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.4**
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	NOT PICS 3/2	
ISUP selection criteria:	PICS 3/8	
Test purpose:	*Overlap procedure supported by determining the end of address signalling. Ti/w1 is expired* Ensure that the SUT in Idle state, on receipt of an IAM message containing Continuity Check indicator in the Nature of Connection Indicators parameter which is set to indicate "**continuity check not required**" on receipt of a SAM start Ti/w1. After Ti/w1 is expired, the SUT: • sends an INVITE message containing all digits received in the IAM and the SAM.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → Start $T_{i/w1}$ SAM → Start $T_{i/w1}$ SAM → Start $T_{i/w1}$ SAM → Start $T_{i/w1}$ $T_{i/w1}$ expired → INVITE ACM(no indication) ← CPG(alerting) ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP302006</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.2.1a</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/Basic call/Receipt of SAM after invite has been sent</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 3/2</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 3/9</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Overlap procedure supported without determining the end of address signalling</i>

Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to indicate "continuity check not required":

• Sends an INVITE message start Ti/w1 and Ti/w2.

• On receipt of a 404/484 the SUT shall send a ACK, stop Ti/w2 and start Ti/w3:

• On receipt of a SAM from the ISUP the SUT shall send an INVITE request:
a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call.
b) All other contents of the new INVITE are interworked from the parameters of the original IAM.
c) Start Ti/w1 and Ti/w2 and stop Ti/w3 (if it is running):</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>ISUP/BICC</th><th></th><th>SUT</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>404/484</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>404/484</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>404/484</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>SAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td>Ringing tone</td><td></td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>
</td></tr>
</table>

6.2.2.3 Sending of the ACM message

TP303001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	PICS 4/9 AND NOT PICS 4/17	
Test purpose:	*ACM is sent after the determination of **address complete** indication in the SUT* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** and the **sending complete** indication: • Sends the INVITE message to called user and starts Ti/w2. • When Ti/w2 is expired, sends the ACM message with the **CPS indicator** set to "no indication (00)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "interworking encountered (1)", the **ISUP indicator** set to "ISUP not used all the way", the **ISDN access indicator** set to "terminating access non-ISDN".	
SIP Parameter values:		
ISUP Parameter values:	IAM; **Called party number**: complete number ACM, **CPS indicator:** no indication (00) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: interworking encountered (1) **ISUP indicator**: ISUP not used all the way **ISDN access indicator:** "terminating access non-ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE Ti/w2 expired ACM(no indication) ← CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 2/3 AND PICS 3/1	
ISUP selection criteria:	PICS 4/9 AND PICS 4/17	
Test purpose:	***64 kBit/s unrestricted*** *call, ACM is sent after the determination of* ***address complete*** *indication in the SUT* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** and the **sending complete** indication: • Sends the INVITE message to called user and starts Ti/w2. • When Ti/w2 is expired, sends the ACM message with the **CPS indicator** set to "no indication (00)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "no interworking encountered (0)", the **ISUP indicator** set to "ISUP used all the way", the **ISDN access indicator** set to "terminating access ISDN".	
SIP Parameter values:	INVITE: SDP a=rtpmap:<dynamic-PT> CLEARMODE/8000	
ISUP Parameter values:	IAM; **Called party number**: complete number, TMR: "64 kbit/s unrestricted" ACM, **CPS indicator:** no indication (00) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: no interworking encountered (0) **ISUP indicator**: ISUP used all the way ISDN access indicator: "terminating access ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE Ti/w2 expired ACM(no indication) ← CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303003	**SIP reference: RFC 3261 [6]**		**ISUP reference: ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5**		
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message				
SIP selection criteria:	PICS 3/1				
ISUP selection criteria:	PICS 4/9 AND NOT PICS 4/17				
Test purpose:	*ACM is sent after the **maximum number of digits** used in the national numbering plan received* Ensure that the SUT in Idle state, on receipt of an IAM message containing the **maximum number of digits used in the national numbering plan**: • Sends the INVITE message to the called user and starts Ti/w2. • When Ti/w2 is expired, sends the ACM message with the **CPS indicator** set to "no indication (00)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "interworking encountered (1)", the **ISUP indicator** set to "ISUP not used all the way", the **ISDN access indicator** set to "terminating access non-ISDN".				
SIP Parameter values:					
ISUP Parameter values:	IAM; **Called party number**: complete number ACM, **CPS indicator:** no indication (00) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: interworking encountered (1) **ISUP indicator**: ISUP not used all the way **ISDN access indicator:** "terminating access non-ISDN"				
Comments:	**ISUP/BICC**		**SUT**		**SIP**
	IAM	→		→	INVITE
			Ti/w2 expired		
	ACM(no indication)	←			
	CPG	←		←	180 Ringing
			Ringing tone		
	ANM	←		←	200 OK INVITE
				→	ACK
			Conversation		
		→		→	BYE
	RLC	←		←	200 OK BYE

TP303004	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 2/3 AND PICS 3/1	
ISUP selection criteria:	PICS 4/9 AND PICS 4/17	
Test purpose:	***64 kBit/s call***, *ACM is sent after the* ***maximum number of digits*** *used in the national numbering plan received* Ensure that the SUT in Idle state, on receipt of an IAM message containing the **maximum number of digits used in the national numbering plan**: • Sends the INVITE message to the called user and starts Ti/w2. • When Ti/w2 is expired, sends the ACM message with the **CPS indicator** set to "no indication (00)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "no interworking encountered", the **ISUP indicator** set to "ISUP used all the way", the **ISDN access indicator** set to "terminating access ISDN".	
SIP Parameter values:	INVITE: SDP a=rtpmap:<dynamic-PT> CLEARMODE/8000	
ISUP Parameter values:	IAM; **Called party number**: complete number ACM, **CPS indicator:** no indication (00) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: no interworking encountered **ISUP indicator**: ISUP used all the way **ISDN access indicator:** "terminating access ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE Ti/w2 expired ACM(no indication) ← CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	PICS 4/9 AND NOT PICS 4/17	
Test purpose:	*ACM is sent after* ***sufficient number of digits*** *has been received to route the call to the called party.* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** where the end of address signalling is determined by analysis of the called party number to indicate that a **sufficient number of digits has been received to route the call to the called party**: • Sends the INVITE message to the called user and starts Ti/w2. • When Ti/w2 is expired, sends the ACM message with the **CPS indicator** set to "no indication (00)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "interworking encountered (1)", the **ISUP indicator** set to "ISUP not used all the way", the **ISDN access indicator** set to "terminating access non-ISDN".	
SIP Parameter values:		
ISUP Parameter values:	IAM; **Called party number**: complete number ACM, **CPS indicator:** no indication (00) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: interworking encountered (1) **ISUP indicator**: ISUP not used all the way **ISDN access indicator:** "terminating access non-ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE Ti/w2 expired ACM(no indication) ← CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP303006</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5</th></tr>
<tr><td>**TSS reference:**</td><td colspan="2">ISUP-SIP /Basic call/Sending of the ACM message</td></tr>
<tr><td>**SIP selection criteria:**</td><td colspan="2">PICS 2/3 AND PICS 3/1</td></tr>
<tr><td>**ISUP selection criteria:**</td><td colspan="2">PICS 4/9 AND PICS 4/17</td></tr>
<tr><td>**Test purpose:**</td><td colspan="2">***64 kBit/s unrestricted call****, ACM is sent after* ***sufficient number of digits*** *has been received to route the call to the called party*

Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** where the end of address signalling is determined by analysis of the called party number to indicate that a **sufficient number of digits has been received to route the call to the called party**:

• Sends the INVITE message to called user and starts Ti/w2.
• When Ti/w2 is expired, sends the ACM message with the **CPS indicator** set to "no indication (00)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "no interworking encountered (0)", the **ISUP indicator** set to "ISUP used all the way", the **ISDN access indicator** set to "terminating access ISDN".</td></tr>
<tr><td>**SIP Parameter values:**</td><td colspan="2">INVITE: SDP a=rtpmap:<dynamic-PT> CLEARMODE/8000</td></tr>
<tr><td>**ISUP Parameter values:**</td><td colspan="2">IAM; **Called party number**: complete number, TMR: "64 kbit/s unrestricted"
ACM, **CPS indicator:** no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)
interworking indicator: no interworking encountered (0)
ISUP indicator: ISUP used all the way
ISDN access indicator: "terminating access ISDN"</td></tr>
<tr><td>**Comments:**</td><td colspan="2">
<table>
<tr><th>ISUP/BICC</th><th></th><th>SUT</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td>Ti/w2 expired</td><td></td><td></td></tr>
<tr><td>ACM</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CPG</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>
</td></tr>
</table>

<table>
<tr><th>TP303007</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP /Basic call/Sending of the ACM message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 3/1</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">NOT PICS 4/17</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>ACM is sent determined by the expiration timer</i> $T_{I/W1}$
Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer $T_{I/W1}$ after the receipt of the latest address message:
• sends the INVITE message to the called user.
• Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM; Called party number: complete number
ACM, CPS indicator: no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)
interworking indicator: interworking encountered (1)
ISUP indicator: ISUP not used all the way
ISDN access indicator: "terminating access non-ISDN"</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>ISUP/BICC</th><th></th><th>SUT</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>→</td><td>Start $T_{I/W1}$</td><td></td><td></td></tr>
<tr><td></td><td></td><td>$T_{I/W1}$ expiry</td><td></td><td></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CPG</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td>Ringing tone</td><td></td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>
</td></tr>
</table>

TP303008	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4 and 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 2/3 AND PICS 3/1	
ISUP selection criteria:	PICS 4/17	
Test purpose:	*64 kBit/s unrestricted call, ACM is sent after determined by the expiration timer* $T_{I/W1}$ Ensure that the SUT in Idle state, on receipt of an IAM message, TMR=64 kBit/s unrestricted containing the complete **called party number** where the end of address signalling is **determined by the expiration timer** $T_{I/W1}$ **after the receipt of the latest address message**: • Sends the INVITE message to called user. • Sends the ACM message with the **CPS indicator** set to "no indication (00)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "no interworking encountered (0)", the **ISUP indicator** set to "ISUP used all the way", the **ISDN access indicator** set to "terminating access ISDN".	
SIP Parameter values:	INVITE: SDP a=rtpmap:<dynamic-PT> CLEARMODE/8000	
ISUP Parameter values:	IAM; **Called party number**: complete number, TMR: "64 kbit/s unrestricted" ACM, **CPS indicator:** no indication (00) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: no interworking encountered (0) **ISUP indicator**: ISUP used all the way **ISDN access indicator:** "terminating access ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → Start $T_{I/W1}$ $T_{I/W1}$ expiry ACM(no indication) ← → INVITE CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP303009</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clauses 7.2.3.2.5 and 7.2.3.2.1.4</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP /Basic call/Sending of the ACM message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 3/2</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">NOT PICS 4/17</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>ACM is sent determined by the expiration timer</i> $T_{\mathbf{I/W2}}$
Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the minimum number of digits required for routing the call has been received (start timer TI/W2 and invoke the appropriate outgoing SIP signalling procedure):
• Sends an INVITE message to the called user and after the expiration of $T_{I/W2.}$
• Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM; Called party number: complete number
ACM, CPS indicator: no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)
interworking indicator: interworking encountered (1)
ISUP indicator: ISUP not used all the way
ISDN access indicator: "terminating access non-ISDN"</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>ISUP/BICC</th><th></th><th>SUT</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>→</td><td>Start $T_{I/W1}$</td><td></td><td></td></tr>
<tr><td>SAM</td><td>→</td><td>Start $T_{I/W1}$</td><td></td><td></td></tr>
<tr><td>SAM</td><td>→</td><td>Start $T_{I/W2}$</td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td>$T_{I/W2}$ expiry</td><td></td><td></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>CPG</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>
</td></tr>
</table>

TP303010	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clauses 7.2.3.2.1.4 and 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 2/3 AND PICS 3/2	
ISUP selection criteria:	PICS 4/17	
Test purpose:	*64 kBit/s unrestricted call, ACM is sent after determined by the expiration timer* $T_{I/W2}$ Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message, TMR=64 kBit/s unrestricted containing the **minimum number of digits required for routing the call has been received** (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure): • Sends an INVITE message to the called user and after the expiration of $T_{OIW2.}$ • Sends the ACM message with the **CPS indicator** set to "no indication (00)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "no interworking encountered (0)", the **ISUP indicator** set to "ISUP used all the way", the **ISDN access indicator** set to "terminating access ISDN".	
SIP Parameter values:	INVITE: SDP a=rtpmap:<dynamic-PT> CLEARMODE/8000	
ISUP Parameter values:	IAM; **Called party number**: complete number, TMR: "64 kbit/s unrestricted" ACM, **CPS indicator:** no indication (00) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: no interworking encountered (0) **ISUP indicator**: ISUP used all the way **ISDN access indicator:** "terminating access ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → Start $T_{I/W1}$ SAM → Start $T_{I/W1}$ SAM → Start $T_{I/W2}$ → INVITE $T_{/IW2}$ expiry ACM(no indication) ← CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303011	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	NOT PICS 4/9 AND NOT PICS 4/17	
Test purpose:	*ACM is sent after 180 Ringing was received* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number,** on receipt of a 180 Ringing message: • Sends the ACM message with the **CPS indicator** set to "subscriber free (01)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "interworking encountered (1)", the **ISUP indicator** set to "ISUP not used all the way", the **ISDN access indicator** set to "terminating access non-ISDN".	
SIP Parameter values:		
ISUP Parameter values:	IAM; **Called party number**: complete number ACM, **CPS indicator:** subscriber free (01) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: interworking encountered (1) **ISUP indicator**: ISUP not used all the way **ISDN access indicator:** "terminating access non-ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303012	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 2/3 AND PICS 3/1	
ISUP selection criteria:	NOT PICS 4/9 AND PICS 4/17	
Test purpose:	*64 kBit/s unrestricted call, ACM is sent after 180 Ringing was received* Ensure that the SUT in Idle state, on receipt of an IAM message, TMR=64 kBit/s unrestricted containing the complete **called party number** on receipt of a 180 Ringing message: • Sends the ACM message with the **CPS indicator** set to "subscriber free (01)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "no interworking encountered (0)", the **ISUP indicator** set to "ISUP used all the way", the **ISDN access indicator** set to "terminating access ISDN".	
SIP Parameter values:	INVITE: SDP a=rtpmap:<dynamic-PT> CLEARMODE/8000	
ISUP Parameter values:	IAM; **Called party number**: complete number, TMR: "64 kbit/s unrestricted" ACM, **CPS indicator:** no indication (00) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: no interworking encountered (0) **ISUP indicator**: ISUP used all the way **ISDN access indicator:** "terminating access ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303013	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.4**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 4/15	
ISUP selection criteria:	NOT PICS 4/9 AND NOT PICS 4/17	
Test purpose:	*The SUT supports the P-Early-Media header* Ensure that the SUT, on receipt of an IAM message containing the complete called party number, where the O-MGCF is supporting the P-Early-Media header as a network option, on the reception of the first 180 Ringing that includes a P-Early-Media header authorizing early media, sends the ACM message with the **CPS indicator** set to " no indication (00)", **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", OBCI "in -band information" set to: yes.	
SIP Parameter values:	180 Ringing: P-Early-Media header	
ISUP Parameter values:	IAM; **Called party number**: complete number ACM; **CPS indicator**: no indication (00), OBCI: in -band information: yes	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303014	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*180 received, mapping of PSTN XML ProgressIindicator #7 into the ACM BCI* Ensure that the SUT, if an ACM has not been already sent, on receipt the 180 Ringing message, with the PSTN XML body with ProgressIindicator # 7 (Terminating user ISDN) • sends the ACM message with the **Called Party"s Status** (CPS) **indicator** set to " subscriber free (01)", the **Called party"s category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "no interworking encountered (0)", the **ISUP indicator** set to "ISUP used all the way" and the **ISDN access indicator** set to "ISDN" and if included the **access delivery information** is set to "Set-up message generated".	
SIP Parameter values:	180 Ringing; PSTN XML body with ProgressIndicator # 7 (Terminating user ISDN)	
ISUP Parameter values:	ACM, **CPS indicator:** subscriber free (01) **Called party"s category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: no interworking encountered (0) **ISUP indicator**: ISUP used all the way **ISDN access indicator:** terminating access isISDN **access delivery information**: Set-up message generated (IF PRESENT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303015	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*180 received, mapping of PSTN XML ProgressIndicator into Progress Indicator contained in the ATP in the sent ACM* Ensure that the SUT, if an ACM has not been already sent, on receipt the 180 Ringing message, with the PSTN XML body with Progress indicator # 7 (Terminating user ISDN) and a PSTN XML ProgressIndicator set to PI_VALUE. The ATP does not contain the ProgressIndicator #7. • sends the ACM message with the **CPS indicator** set to " subscriber free (01)" and the **Access Transport Parameter** (ATP) containing the **progress indicator** PI_VALUE.	
SIP Parameter values:	180 Ringing; PSTN XML **ProgressIndicator**: PI_VALUE (PIXIT)	
ISUP Parameter values:	ACM, CPS indicator: subscriber free (01) ATP progress indicator: PI_VALUE (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

Values and additional selection criteria for test purposes TP303015	
VA_01	PI_VALUE = Call is not end-to-end ISDN (#1)
VA_02	PI_VALUE = Destination address is non-ISDN (#2)

TP303016	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 3/1 AND NOT PICS 4/15	
ISUP selection criteria:	NOT PICS 4/9 AND NOT PICS 4/17	
Test purpose:	*P-Early-Media header not supported, 183 is not interworked sending complete indication received* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** and the **sending complete** indication, on receipt of a 183 Session Progress: • Sends the INVITE message to called user. • No ISUP message is sent backward.	
SIP Parameter values:		
ISUP Parameter values:	IAM; **Called party number**: complete number	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 183 Session Progress ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303017	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 4/15	
ISUP selection criteria:		
Test purpose:	*P-Early-Media header supported, 183 is interworked, an ACM no indication is sent* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number, where the O-MGCF is supporting the P-Early-Media header if the 183 Session Progress contains a P-Early_Media header authorizing early media • sends the ACM message with the **CPS indicator** set to " no indication (00)", the **Called party"s category indicator** set to "no indication(00)", OBCI "in -band information" set to: yes.	
SIP Parameter values:	183 Session Progress that includes a P-Early-Media header authorizing early media	
ISUP Parameter values:	IAM; **Called party number**: complete number ACM; **CPS indicator**: no indication (00), OBCI: in -band information: yes	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM(no indication) ← ← 183 Session Progress CPG(Alerting) ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303018	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 9.2.3.3.12**
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	NOT PICS 4/15	
Test purpose:	*P-Early-Media header not supported, 183 is not interworked maximum number of digits used in the national numbering plan received* Ensure that the SUT in Idle state, on receipt of an IAM message containing the **maximum number of digits used in the national numbering plan** on receipt of a 183 Session Progress: • No ISUP message is sent backward.	
SIP Parameter values:		
ISUP Parameter values:	IAM; **Called party number**: complete number	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 183 Session Progress ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303019	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 9.2.3.3.12**
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	NOT PICS 4/15	
Test purpose:	*P-Early-Media header not supported, 183 is not interworked sufficient number of digits has been received to route the call to the called party received* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** where the end of address signalling is determined by analysis of the called party number to indicate that a **sufficient number of digits has been received to route the call to the called party** on receipt of a 183 Session Progress: • No BICC/ISUP message is sent backward.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 183 Session Progress ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303020	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 9.2.3.3.12**
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 3/1 NOT PICS 4/15	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	*183 received after $T_{I/W1}$ expired, P-Early-Media header not supported 183 is not interworked* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number** where the end of address signalling is **determined by the expiration timer $T_{I/W1}$** after the receipt of the latest address message on receipt of a 183 Session Progress: • No ISUP message is sent backward.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → $T_{I/W1}$ expiry ACM(no indication) ← → INVITE ← 183 Session Progress CPG(alerting) ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303021	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 4/15 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*183 received, mapping of PSTN XML ProgressIindicator #7 into the ACM BCI* Ensure that the SUT, on receipt of the 183 Session Progress message with the **PSTN XML ProgressIndicator** # 7 (Terminating user ISDN) • sends the ACM message with the **CPS indicator** set to " no indication (00)", the **Called party"s category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "no interworking encountered (0)", the **ISUP indicator** set to "ISUP used all the way" and the **ISDN access indicator** set to "ISDN" and if included the **access delivery information** is set to "Set-up message generated".	
SIP Parameter values:	183 Session Progress; **PSTN XML ProgressIndicator** # 7 (Terminating user ISDN)	
ISUP Parameter values:	ACM, **CPS indicator:** no indication (00) **Called party"s category indicator:** no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator:** no interworking encountered (0) **ISUP indicator:** ISUP used all the way **ISDN access indicator:** ISDN **access delivery information:** Set-up message generated (IF PRESENT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 183 Session Progress CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP03022	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 4/15 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*183 received, mapping of PSTN XML ProgressIndicator into Progress Indicator contained in the ATP in the sent ACM* Ensure that the SUT, on receipt of the 183 Session Progress message with the PSTN XML body with Progress indicator # 7 (Terminating user ISDN) containing the **PSTN XML ProgressIndicator** set to PI_VALUE. The XML ProgressIndicator #7 is not interworked. • sends the ACM message with the **CPS indicator** set to " no indication (00)", the **Called party"s category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "no interworking encountered (0)", the **ISUP indicator** set to "ISUP used all the way", the **ISDN access indicator** set to "ISDN", the **Access Transport Parameter** (ATP) containing the **progress indicator** set to PI_VALUE and if included the **access delivery information** is set to "Set-up message generated".	
SIP Parameter values:	183 Session Progress; **PSTN XML ProgressIndicator**: PI_VALUE	
ISUP Parameter values:	183 Session Progress, **CPS indicator:** no indication (00) **Called party"s category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: no interworking encountered (0) **ISUP indicator**: ISUP used all the way **ISDN access indicator:** ISDN **ATP progress indicator**: PI_VALUE **access delivery information**: Set-up message generated (IF PRESENT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM ➔ ➔ INVITE ACM ← ← 183 Session Progress CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE ➔ ACK Conversation ➔ ➔ BYE RLC ← ← 200 OK BYE	

Values for test purposes TP303022	
VA_01	PI_VALUE: Call is not end-to-end ISDN: further call progress information is available in-band (#1)
VA_02	PI_VALUE: Destination address is non-ISDN (#2)

TP303023	• **SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	NOT PICS 4/15 AND NOT PICS 4/18	
ISUP selection criteria:		
Test purpose:	*183 received, mapping of PSTN XML ProgressIndicator into Progress Indicator contained in the ATP and mapping of PSTN XML ProgressIindicator #7 BCIis not supported* Ensure that the SUT, on receipt of the 183 Session Progress message with the PSTN XML body with Progress indicator # 7 (Terminating user ISDN) and containing the **PSTN XML ProgressIndicator** set to PI_VALUE, • does not send the ACM message.	
SIP Parameter values:	183 Session Progress; **PSTN XML ProgressIndicator**: PI_VALUE	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 183 Session Progress ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

Values for test purposes TP303023	
VA_01	PI_VALUE: originating address is non-ISDN (#3)
VA_02	PI_VALUE: Call has returned to ISDN (#4)

TP303024	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.3.3.2.3**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 3/1 AND PICS 4/5 AND PICS 4/11	
ISUP selection criteria:	PICS 4/2 AND PICS 4/9 AND NOT PICS 4/17	
Test purpose:	*Preconditions requested, ACM is sent after the determination of address complete indication in the SUT* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number,** the **sending complete** indication, and the continuity check is required on this circuit (ISUP) or COT is expected (BICC): • Sends the INVITE message to called user and starts Ti/w2. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the **CPS indicator** set to "no indication (00)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "interworking encountered (1)", the **ISUP indicator** set to "ISUP not used all the way", the **ISDN access indicator** set to "terminating access non-ISDN".	
SIP Parameter values:		
ISUP Parameter values:	IAM; **Called party number:** complete number ACM, **CPS indicator:** no indication (00) **Called party's category indicator:** no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: interworking encountered (1) **ISUP indicator:** ISUP used all the way **ISDN access indicator:** "terminating access non-ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 183 Session Progress → PRACK ← 200 OK PRACK COT → → UPDATE ← 200 OK UPDATE ACM ← Ti/w2 expired CPG(Alerting) ← ← 180 Ringing → PRACK ← 200 OK PRACK Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303025	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.3.3.2.3**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 3/1 AND PICS 4/5 AND PICS 4/11	
ISUP selection criteria:	PICS 4/2 AND PICS 4/9 AND PICS 4/17	
Test purpose:	*64 kBit/s, Preconditions requested, ACM is sent after the determination of address complete indication in the SUT* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number,** the **sending complete** indication and the continuity check is required on this circuit (ISUP) or COT is expected (BICC): • Sends the INVITE message to called user and starts Ti/w2. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the **CPS indicator** set to "subscriber free (01)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "no interworking encountered (0)", the **ISUP indicator** set to "ISUP used all the way", the **ISDN access indicator** set to "terminating access ISDN".	
SIP Parameter values:	INVITE: SDP a=rtpmap:<dynamic-PT> CLEARMODE/8000	
ISUP Parameter values:	IAM; **Called party number**: complete number, TMR: "64 kbit/s unrestricted" ACM, **CPS indicator:** no indication (00) **Called party's category indicator**: no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator**: no interworking encountered (0) **ISUP indicator**: ISUP used all the way **ISDN access indicator:** "terminating access ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 183 Session Progress → PRACK ← 200 OK PRACK COT → → UPDATE ← 200 OK UPDATE ACM(no indication) ← Ti/w2 expired CPG ← ← 180 Ringing → PRACK ← 200 OK PRACK ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303028	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.3.3.2.3**
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 3/2 AND PICS 4/5 AND PICS 4/11	
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/17	
Test purpose:	*Preconditions requested, ACM is sent after expiration of timer* $T_{I/W2}$ Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the **minimum number of digits required for routing the call** has been received, start timer TI/W2 and invoke the appropriate outgoing SIP signalling procedure and the continuity check is required on this circuit (ISUP): • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the **CPS indicator** set to "subscriber free (01)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "interworking encountered (1)", the **ISUP indicator** set to "ISUP not used all the way", the **ISDN access indicator** set to "terminating access non ISDN".	
SIP Parameter values:	INVITE: SDP a=rtpmap:<dynamic-PT> CLEARMODE/8000	
ISUP Parameter values:	IAM; **Called party number:** complete number ACM, **CPS indicator:** no indication (00) **Called party's category indicator:** no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator:** interworking encountered (1) **ISUP indicator:** ISUP not used all the way **ISDN access indicator:** "terminating access non ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 183 Session Progress → PRACK ← 200 OK PRACK COT → → UPDATE ← 200 OK UPDATE $T_{I/W2}$ expiry ACM(no indication) ← CPG(Alerting) ← ← 180 Ringing → PRACK ← 200 OK PRACK Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303029	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.3.3.2.3**
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 2/3 AND PICS 3/2 AND PICS 4/5 AND PICS 4/11	
ISUP selection criteria:	PICS 4/2 AND PICS 4/17	
Test purpose:	*64 kBit/s call, Preconditions requested, ACM is sent after expiration of timer* $T_{I/W2}$ Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the **minimum number of digits required for routing the call** has been received, start timer TI/W2 and invoke the appropriate outgoing SIP signalling procedure and the continuity check is required on this circuit (ISUP): • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the **CPS indicator** set to "subscriber free (01)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "no interworking encountered (0)", the **ISUP indicator** set to "ISUP used all the way", the **ISDN access indicator** set to "terminating access ISDN".	
SIP Parameter values:	INVITE: SDP a=rtpmap:<dynamic-PT> CLEARMODE/8000	
ISUP Parameter values:	IAM; **Called party number:** complete number, TMR: "64 kbit/s unrestricted" ACM, **CPS indicator:** no indication (00) **Called party's category indicator:** no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator:** no interworking encountered (0) **ISUP indicator:** ISUP used all the way **ISDN access indicator:** "terminating access ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 183 Session Progress → PRACK ← 200 OK PRACK COT → → UPDATE ← 200 OK UPDATE $T_{I/W2}$ expiry ACM ← CPG ← ← 180 Ringing → PRACK ← 200 OK PRACK Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303030	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.3.3.2.3**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 4/5 AND PICS 4/11	
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9 AND NOT PICS 4/17	
Test purpose:	*180 received after precontitions met, an ACM is sent* Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete **called party number,** the continuity check is required on this circuit (ISUP) indication receipt of a 180 Ringing message: • Sends the ACM message with the **CPS indicator** set to "subscriber free (01)", the **Called party's category indicator** set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the **interworking indicator** set to "interworking encountered (1)", the **ISUP indicator** set to "ISUP not used all the way", the **ISDN access indicator** set to "terminating access non-ISDN" .	
SIP Parameter values:		
ISUP Parameter values:	IAM; **Called party number:** complete number ACM, **CPS indicator:** "subscriber free (01)" **Called party's category indicator:** no indication(00) or ordinary subscriber (01) or payphone (10) **interworking indicator:** interworking encountered (1) **ISUP indicator:** ISUP used all the way **ISDN access indicator:** "terminating access non-ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 183 Session Progress → PRACK ← 200 OK PRACK COT → → UPDATE ← 200 OK UPDATE ACM ← ← 180 Ringing → PRACK ← 200 OK PRACK Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP303031</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.3.3.2.3</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP /Basic call/Sending of the ACM message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 2/3 AND PICS 4/5 AND PICS 4/11</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 4/2 AND NOT PICS 4/9 AND PICS 4/17</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>64 kBit/s call, 180 received after precontitions met, an ACM is sent</i>
Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number, the continuity check is required on this circuit (ISUP) or COT is expected (BICC) indication on receipt of a 180 Ringing message:
• Sends the ACM message with theCPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "no interworking encountered (0)", the ISUP indicator set to "ISUP used all the way", the ISDN access indicator set to "terminating access ISDN".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INVITE: SDP a=rtpmap:<dynamic-PT> CLEARMODE/8000</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM; Called party number: complete number, TMR: "64 kbit/s unrestricted"
ACM, CPS indicator: no indication (00)
Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)
interworking indicator: no interworking encountered (0)
ISUP indicator: ISUP used all the way
ISDN access indicator: "terminating access ISDN"</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>ISUP/BICC</th><th></th><th>SUT</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>183 Session Progress</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td>COT</td><td>→</td><td></td><td>→</td><td>UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>PRACK</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK PRACK</td></tr>
<tr><td></td><td colspan="2">Ringing tone</td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>
</td></tr>
</table>

TP303032	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 2/3 AND PICS 4/18 AND PICS 4/19	
ISUP selection criteria:		
Test purpose:	*Mapping of PSTN XML BearerCapability element contained in the 180 into the TMU parameter sent in the ACM* Ensure on receipt of a 180 Ringing containes PSTN XML ProgressIndicator #7 and PSTN XML BearerCapability BC_VALUE, an ACM is sent containing the **TMU Parameter BC_VALUE**. The BCI is set to: ISUP indicator: ISUP is used all the way ISDN access indicator: ISDN Interwoking indicator: Interworking not enountered	
SIP Parameter values:	INVITE; **PSTN XML first Bearer Capability:** INVITE _BC1 **PSTN XML second Bearer Capability:** INVITE _BC2 180 Ringing PSTN XML BC: BC_VALUE and XML PI #7	
ISUP Parameter values:	IAM; **USI:** Speech/audio3Kbit/s, G.711 A-law **USI** prime: Unrestr. Digital info T/A, G.711 A-law **TMR:** 64 kbit/s preferred **TMR prime:** Speech/audio3Kbit/s **ACM:** ISUP indicator: ISUP is used all the way ISDN access indicator: ISDN Interwoking indicator: Interworking not enountered TMU: BC_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP303033	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.5**
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 2/3 AND PICS 4/18 AND PICS 4/19	
ISUP selection criteria:		
Test purpose:	*Mapping of PSTN XML PearerCapability element contained in the 183 into the TMU parameter sent in the ACM* Ensure on receipt of a 183 Session Progress containes PSTN XML ProgressIndicator #7 and PSTN XML BearerCapability BC_VALUE, an ACM is sent containing the **TMU Parameter BC_VALUE**. The BCI is set to: ISUP indicator: ISUP is used all the way ISDN access indicator: ISDN Interwoking indicator: Interworking not enountered	
SIP Parameter values:	INVITE; **PSTN XML first Bearer Capability:** INVITE _BC1 **PSTN XML second Bearer Capability:** INVITE _BC2 183 Session Progress PSTN XML BC: BC_VALUE and XML PI #7	
ISUP Parameter values:	IAM; **USI:** Speech/audio3Kbit/s, G.711 A-law **USI** prime: Unrestr. Digital info T/A, G.711 A-law **TMR:** 64 kbit/s preferred **TMR prime:** Speech/audio3Kbit/s **ACM:** ISUP indicator: ISUP is used all the way ISDN access indicator: ISDN Interwoking indicator: Interworking not enountered TMU: BC_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM(no indication) ← ← 183 Session Progress ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

Values and selection criteria for test purpose TP303032a and TP303033			
Test purposes	**ACM Parameter values**	**18x Provisional response values:**	**INVITE parameter value**
VA_01	TMU_VALUE: speech	PSTN XML: BC_VALUE: speech	PSTN XML INVITE BC1: speech INVITE BC2: unrestricted digital information with tones and announcements
VA_02	TMU_VALUE: 3,1 kHz	PSTN XML: BC_VALUE: 3,1 kHz audio	PSTN XML INVITE BC1: 3,1 kHz audio INVITE BC2: unrestricted digital information with tones and announcements

6.2.2.4 Sending of the CPG message

TP304001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.6**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	PICS 4/9	
Test purpose:	*180 received, a CPG is sent when an ACM was sent before* Ensure that the SUT, having sent a ACM message with called party status "no indication" on receipt of a 180 Ringing message: - Sends the CPG message with the with the **event indicator** set to "Alerting".	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE Ti/w2 expired ACM(no indication) ← CPG(Alerting) ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP304002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.6**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	NOT PICS 4/15	
Test purpose:	*ACM was sent after $T_{I/W1}$ expiry, a 183 is not interworked* Ensure that the SUT, having sent a ACM message with called party status "no indication" after $T_{I/W1}$ expiry, on receipt of a 183 Session progress message: - ISUP message is sent backward.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → $T_{I/W1}$ expiry ACM ← → INVITE ← 183 Session Progress Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP304003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:	PICS 3/1 AND PICS 4/18 AND PICS 4/19	
ISUP selection criteria:		
Test purpose:	*ACM was sent after $T_{I/W1}$ expiry, a CPG is sent when a 183 is received contains a PSTN XML ProgressIndicator #7* Ensure that the SUT, having sent the ACM message after the expiry of ToiW1, on receipt of 183 Session progress message, with **PSTN XML ProgressIndicator** "*Terminating access ISDN"(#7)* • sends the CPG message with the **event indicator** set to "progress".	
SIP Parameter values:	183 Session Progress ; **PSTN XML ProgressIndicator** "*Terminating access ISDN"(#7)*	
ISUP Parameter values:	CPG; **event indicator:** progress BCI **interworking indicator:** no interworking encountered (0) **ISUP indicator:** ISUP used all the way ISDN access indicator: "terminating access ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → $T_{I/W1}$ expiry ACM ← → INVITE CPG(Progress) ← ← 183 Session progress CPG(Alerting) ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP304004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:	PICS 3/1 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ACM was sent after, a 183 contains a PSTN XML ProgressIndicator #7 was received* Ensure that the SUT, having sent the ACM message after the reception of the 183 Session progress message, with **PSTN XML ProgressIndicator** "*Terminating access ISDN"(#7)*, on receipt of an 180 Ringing message, with PSTN XML body containing the progress descriptions "*Terminating access ISDN"(#7)* • sends the CPG message with the **event indicator** set to "Alerting".	
SIP Parameter values:	183 Session Progress: **ProgressIndicator** "*Terminating access ISDN"(#7)* 180 Ringing: **ProgressIndicator** "*Terminating access ISDN"(#7)*	
ISUP Parameter values:	CPG; **event indicator:** Alerting BCI **interworking indicator:** no interworking encountered (0) **ISUP indicator**: ISUP used all the way ISDN access indicator: "terminating access ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 183 Session progress CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP304005</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
ES 283 027 [1], clause 7.2.3.1.4</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP /Basic call/ Sending of the CPG message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 3/1 AND PICS 4/18</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>ACM was sent after</i> $T_{I/W1}$ expiry<i>, after receipt of a 183 and 180 contains a PSTN XML ProgressIndicator #7 a CPG(Progress) and a CPG(Alerting) are sent</i>
Ensure that the SUT, having sent the ACM message after the expiry of Ti/W1, on receipt of a 183 Session progress message followed by an 180 Ringing message with PSTN XML body containing the progress descriptions "<i>Terminating access ISDN"(#7)</i>,
• sends two CPG messages respectively with the event indicator set to "Progress" and "Alerting".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">183 Session progress
180 Ringing</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5">CPG 1; event indicator: Progress
CPG 2; event indicator: Alerting
BCI:
interworking indicator: no interworking encountered
ISUP indicator: ISUP used all the way
ISDN access indicator: "terminating access ISDN"</td></tr>
<tr><td rowspan="12">Comments:</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>$T_{I/W1}$ expiry</td><td></td><td></td></tr>
<tr><td>ACM(no indication)</td><td>←</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>CPG(Progress)</td><td>←</td><td></td><td>←</td><td>183 Session progress</td></tr>
<tr><td>CPG(Alerting)</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td colspan="3">Ringing tone</td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>

TP304006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:	PICS 3/1 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ACM was sent after $T_{I/W1}$ expiry, a 183 covering a PSTN XML ProgressIndicator #7 and #x received, a CPG is sent contains an ATP with PI #x* Ensure that the SUT, having sent the ACM message after the expiry of Ti/W1, on receipt of 183 Session progress message, with PSTN XML body containing the progress descriptions "*Terminating access ISDN"(#7) and* PI_VALUE the ProgressIndicator #7 is not sent in the ATP. • sends the CPG message with the **event indicator** set to "progress" and the **ATP progress indicator** set to PI_VALUE .	
SIP Parameter values:	183 Session Progress ; PSTN XML ProgressIndicator "*Terminating access ISDN"(#7)* PI_VALUE	
ISUP Parameter values:	CPG; **event indicator**: progress **ATP progress indicator**: PI_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → $T_{I/W1}$ expiry ACM(no indication) ← → INVITE CPG(Progress) ← ← 183 Session progress CPG(Alerting) ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

Values and additional selection criteria for test purposes TP304006	
VA_01	PI_VALUE = Call is not end-to-end ISDN (#1)
VA_02	PI_VALUE = Destination address is non-ISDN (#2)

TP304007	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:	PICS 3/1 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ACM was sent after $T_{I/W1}$ expiry, a 180 covering a PSTN XML ProgressIndicator #7 and #x received, a CPG is sent contains an ATP with PI #x* Ensure that the SUT, having sent the ACM message, on receipt of an a 180 Ringing message with **PSTN XML ProgressIndicator** "*Terminating access ISDN"(#7)"* and " PI_VALUE ", • sends a CPG message with the **event indicator** set to "Alerting" and the **ATP** including the **progress indicator** set to " PI_VALUE".	
SIP Parameter values:	180 Ringing; **PSTN XML ProgressIndicator**: PI_VALUE	
ISUP Parameter values:	CPG; **Event indicator**: Alerting **ATP progress indicator**: PI_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM ➔ T_{OIW1} expiry ACM(no indication) ← ➔ INVITE CPG(Alerting) ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE ➔ ACK Conversation ➔ ➔ BYE RLC ← ← 200 OK BYE	

Values and additional selection criteria for test purposes TP304007	
VA_01	PI_VALUE = Call is not end-to-end ISDN (#1)
VA_02	PI_VALUE = Destination address is non-ISDN (#2)

TP304008	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:	PICS 2/3 AND PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*ACM was sent after $T_{I/W1}$ expiry, a 180 covering a PSTN XML ProgressIndicator #7 and #x received, a CPG is sent contains an ATP with PI #x* Ensure that the SUT having sent the ACM message, on receipt of a 183 Call Progress message containing the **PSTN XML ProgressIndicator** "*Terminating access ISDN"(#7) and* "Interworking has occurred and has resulted in a telecommunication service change (#5)"*, the* PSTN XML **BearerCapability** set to BC_VALUE • sends the CPG message with the **event indicator** set to "Progress", the **ATP** containing the **BC** set to BC_VALUE and the **progress indicator** set to "Interworking has occurred and has resulted in a telecommunication service change (#5)" and the **TMU** set to TMU_VALUE.	
SIP Parameter values:	INVITE; **PSTN XML first Bearer Capability**: INVITE _BC1 **PSTN XML second Bearer Capability**: INVITE _BC2 183 Call Progress; **PSTN XML ProgressIndicator**: Interworking has occurred and has resulted in a telecommunication service change (#5). PSTN XML **BearerCapability**: BC_VALUE	
ISUP Parameter values:	IAM; **USI**: Speech/audio3Kbit/s, G.711 A-law **USI** prime: Unrestr. Digital info T/A, G.711 A-law **TMR**: 64 kbit/s preferred **TMR prime**: Speech/audio3Kbit/s CPG, **event indicator:** Progress **ATP BC**: BC_VALUE **ATP progress indicator**: Interworking has occurred and has resulted in a telecommunication service change (#5) **TMU**: TMU_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → T_{OIW1} expiry ACM ← → INVITE CPG ← ← 183 Session progress CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP304009	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:	PICS 2/3 AND PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*ACM was sent after $T_{I/W1}$ expiry, Fallback occurs in the 180 Ringing* Ensure that the SUT in call having sent the ACM message, on receipt of an 180 Ringing message containing the **PSTN XML BearerCapability** set to BC_VALUE and the **PSTN XML ProgressIndicator** set to "*Terminating access ISDN"(#7) and*"Interworking has occurred and has resulted in a telecommunication service change (#5)" • sends the CPG message with the **event indicator** set to "Alerting", the **ATP** containing the **BC** set to BC_VALUE and the **progress indicator** set to "Interworking has occurred and has resulted in a telecommunication service change (#5)" and the **TMU** set to TMU_VALUE.	
SIP Parameter values:	INVITE; **PSTN XML first Bearer Capability**: INVITE _BC1 **PSTN XML second Bearer Capability**: INVITE _BC2 180 Ringing; **PSTN XML ProgressIndicator**: "*Terminating access ISDN"(#7) and* Interworking has occurred and has resulted in a telecommunication service change (#5) **PSTN XML BeaereCapabilty**: BC_VALUE	
ISUP Parameter values:	IAM; **USI**: Speech/audio3Kbit/s, G.711 A-law **USI** prime: Unrestr. Digital info T/A, G.711 A-law **TMR**: 64 kbit/s preferred **TMR prime**: Speech/audio3Kbit/s CPG; **event indicator**: Alerting **ATP BC**: BC_VALUE **ATP progress indicator**: Interworking has occurred and has resulted in a telecommunication service change (#5) **TMU**: TMU_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → T_{OIW1} expiry ACM ← → INVITE CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP304010	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4.1**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:	PICS 2/3 AND PICS 4/18	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*ACM was sent after* $T_{I/W1}$ *expiry, Fallback occurs in the 183 Session Progress* Ensure that the SUT having sent the ACM message, on receipt of a 183 Session Progress message containing the **BC** SET to BC_VALUE and the **PSTN XML ProgressIndicator** set to "*Terminating access ISDN"(#7),* "Interworking has occurred and has resulted in a telecommunication service change (#5)" and "In-band information or appropriate pattern is now available (#8)", • sends the CPG message with the **event indicator** set to "In-band information or appropriate pattern is now available", the **ATP** containing the **BC** set to BC_VALUE and the **progress indicator** set to "Interworking has occurred and has resulted in a telecommunication service change (#5)" and the **TMU** set to TMU_VALUE.	
SIP Parameter values:	INVITE; **PSTN XML first Bearer Capability**: INVITE _BC1 **PSTN XML second Bearer Capability**: INVITE _BC2 183 Session Progress ; **PSTN XML** **ProgressIndicator**: Interworking has occurred and has resulted in a telecommunication service change (#5) **ProgressIndicator**: In-band information or appropriate pattern is now available (#8) **PSTN XML BeaererCapability**: BC_VALUE	
ISUP Parameter values:	IAM; **USI**: Speech/audio3Kbit/s, G.711 A-law **USI** prime: Unrestr. Digital info T/A, G.711 A-law **TMR**: 64 kbit/s preferred **TMR prime**: Speech/audio3Kbit/s CPG; **event indicator**: In-band information or appropriate pattern is now available **ATP BC**: BC_VALUE **ATP progress indicator**: Interworking has occurred and has resulted in a telecommunication service change (#5) **TMU**: TMU_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → T_{OIW1} expiry ACM ← → INVITE CPG ← ← 183 Session progress CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

Values and additional selection criteria for test purposes TP304008 to TP304010		
VA_01	TMU_VALUE: speech ISUP_VALUE: UDI/TA	BC_VALUE: speech
VA_02	TMU_VALUE: 3,1 kHz ISUP_VALUE: UDI/TA	BC_VALUE: 3,1 kHz

TP3040011	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4.1**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*ACM was sent after* $T_{I/W1}$ *expiry, PSTN XML HLC received in a 183 mapping in the ATP contained in the CPG* Ensure that the SUT, having sent the ACM message, on receipt of a 183 Session Progress message with **PSTN XML ProgressIndicator** set to "Interworking has occurred and has resulted in a telecommunication service change (#5)", "*Terminating access ISDN"(#7)* and with a **PSTN XML HighLayerCompatibility** set to HLC_VALUE the ProgressIndicator #7 is not contained in the ATP • sends the CPG message with **event indicator** set to "Progress", the **ATP** including the HLC set to HLC_VALUE and the **progress indicator** set to "Interworking has occurred and has resulted in a telecommunication service change (#5)".	
SIP Parameter values:	183 Session Progress; **PSTN XML ProgressIndicator**: Interworking has occurred and has resulted in a telecommunication service change (#5) and "*Terminating access ISDN"(#7)* **PSTN XML HighLayerCompatibility**: HLC_VALUE2 (PIXIT)	
ISUP Parameter values:	CPG, **Event indicator**: Progress **ATP HLC**: HLC_VALUE2 (PIXIT) **ATP progress indicator**: Interworking has occurred and has resulted in a telecommunication service change (#5)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → T_{OIW1} expiry ACM ← → INVITE CPG ← ← 183 Session progress CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP3040012</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.4</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP /Basic call/ Sending of the CPG message</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 4/18</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>ACM was sent after $T_{I/W1}$ expiry, PSTN XML HLC received in a 180 mapping in the ATP contained in the CPG</i>
Ensure that the SUT, having sent the ACM message, on receipt of an 180 Ringing message with PSTN XML ProgressIndicator set to "Interworking has occurred and has resulted in a telecommunication service change (#5)" "<i>Terminating access ISDN"(#7)</i>and with a PSTN XML HighLayerCompatibility set to HLC_VALUE
• sends the CPG message with event indicator set to "Alert", the ATP including the HLC set to HLC_VALUE and the progress indicator set to "Interworking has occurred and has resulted in a telecommunication service change (#5)".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">180 Ringing;
PSTN XML ProgressIndicator: Interworking has occurred and has resulted in a telecommunication service change (#5), "<i>Terminating access ISDN"(#7)</i>
PSTN XML HighLayerCompatibility: HLC_VALUE2 (PIXIT)</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">CPG, Event indicator: Alerting
ATP HLC: HLC_VALUE2 (PIXIT)
ATP progress indicator: Interworking has occurred and has resulted in a telecommunication service change (#5)</td></tr>
<tr><td>Comments:</td><td colspan="2">ISUP/BICC SUT SIP
IAM →
T_{OIW1} expiry
ACM ← → INVITE
CPG ← ← 180 Ringing
Ringing tone
ANM ← ← 200 OK INVITE
→ ACK
Conversation
→ → BYE
RLC ← ← 200 OK BYE</td></tr>
</table>

TP3040013	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:		
ISUP selection criteria:	PICS 4/9	
Test purpose:	*ACM sent after INVITE was sent, 183 received after 180 received. 183 contains a PSTN XML ProgressIndicator #7 mapped into BCI in the sent CPG* Ensure that the SUT, having sent automatically the ACM message, on receipt of an 180 Ringing message followed by a 183 Session Progress message with **PSTN XML ProgressIndicator** "*Terminating access ISDN"(#7)*, • sends two CPG message respectively with the **event indicator** set to "Alerting" and "Progress".	
SIP Parameter values:	180 Ringing; 183 Session Progress ;	
ISUP Parameter values:	CPG 1; **event indicator**: Alerting ISUP indicator: ISUP is used all the way ISDN access indicator: ISDN Interwoking indicator: Interworking not enountered CPG 2; **event indicator**: Progress ISUP indicator: ISUP is used all the way ISDN access indicator: ISDN Interwoking indicator: Interworking not enountered	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → ACM ← → INVITE CPG ← ← 180 Ringing CPG ← ← 183 Session progress Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP3040014	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:		
ISUP selection criteria:	PICS 4/9	
Test purpose:	*ACM sent after INVITE was sent, a 180 is received contains a PSTN XML ProgressIndicator #x mapped into an PI #x covered in an ATP in the sent CPG* Ensure that the SUT, having sent automatically the ACM message, on receipt of an 180 Ringing message containing the **PSTN XML ProgressIndicator** set to PI_VALUE, • sends a CPG message with the **event indicator** set to "Alerting" and the **ATP** including the **progress indicator** set to PI_VALUE.	
SIP Parameter values:	180 Ringing; **progress indicator**: PI_VALUE	
ISUP Parameter values:	CPG; **Event indicator**: Alerting **ATP progress indicator**: PI_VALUE	
Comments:	**SUT** **SIP** IAM → ACM ← → INVITE CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Values and additional selection criteria for test purposes TP3040014	
VA_01	PI_VALUE = Call is not end-to-end ISDN (#1)
VA_02	PI_VALUE = Destination address is non-ISDN (#2)

TP3040015	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.4**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:		
ISUP selection criteria:	PICS 4/9	
Test purpose:	*ACM sent after INVITE was sent, 180 received contains a PSTN XML ProgressIndicator #8 mapped into OBCI in the sent CPG* Ensure that the SUT, having received an IAM with the **USI field** indicating USI_VALUE and having sent automatically the ACM message, on receipt of an 180 Ringing message with PI No.8 "In-band information or appropriate pattern is now available", • sends a CPG message with the **event indicator** set to "Alerting" and **OBCI in-band** information set to "yes".	
SIP Parameter values:	180 Ringing; PSTN XML ProgressIndicator"In-band information or appropriate pattern is now available" (#8)	
ISUP Parameter values:	IAM: **USI**: USI_VALUE; CPG; **Event indicator**: Alerting **OBCI in-band**: yes	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → ACM ← → INVITE CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

TP3040016	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause**
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:		
ISUP selection criteria:	PICS 4/9	
Test purpose:	*ACM sent after INVITE was sent, 180 rceived contains a P-Early-Media header mapped into the OBCI "inband info available"* Ensure that the SUT, having received an IAM with the **USI field** indicating USI_VALUE and having sent automatically the ACM message, on receipt of an 180 Ringing message with P-Early-Media header authorizing early media", • sends a CPG message with the **event indicator** set to "Alerting" and **OBCI in-band** information set to "yes".	
SIP Parameter values:	180 Ringing; P-Early-Media header authorizing early media	
ISUP Parameter values:	IAM: **USI**: USI_VALUE; CPG; **Event indicator**: Alerting **OBCI in-band**: yes	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → ACM ← → INVITE CPG ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation → → BYE RLC ← ← 200 OK BYE	

Values and additional selection criteria for test purposes TP304016	
VA_01	USI_VALUE = speech
VA_02	USI_VALUE = 3,1 kHz

6.2.2.5 Sending of the ANM message

TP305001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.7a**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*An ANM is sent after a 200 OK INVITE is received* Ensure that the SUT having sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running): • Send ANM as determined by BICC/ISUP procedures. • Stop any existing awaiting answer indication (e.g. ringing tone).	
SIP Parameter values:	200 OK INVITE;	
ISUP Parameter values:	ANM;	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP305002	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*An ANM is sent after a 200 OK INVITE is received. PSTN XML ProgressIndicator #x mapped into the ATP in the ANM* Ensure that the SUT, having sent the ACM message, on receipt of a 200 OK message containing the **PSTN XML ProgressIndicator** set to PI_VALUE • sends the ANM message with the **ATP** including the **PSTN XML ProgressIndicator** set to PI_VALUE.	
SIP Parameter values:	200 OK; **PSTN XML ProgressIndicator**: PI_VALUE (PIXIT)	
ISUP Parameter values:	ANM; **ATP Progress Indicator**: PI_VALUE (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP305003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*An ANM is sent after a 200 OK INVITE is received. PSTN XML LowLayerCompatibility mapped into the ATP in the ANM* Ensure that the SUT, having sent the ACM message, on receipt of a 200 OK Message containing the **PSTN XML LowLayerCompatibility** set to LLC_VALUE • sends the ANM message with the **ATP** including the **LLC** set to LLC_VALUE.	
SIP Parameter values:	200 OK; **PSTN XML LowLayerCompatibility**: LLC_VALUE (PIXIT)	
ISUP Parameter values:	ANM; **ATP LLC**: PI_VALUE (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP305004	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/	
SIP selection criteria:	PICS 4/18 AND PICS 2/3	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*An ANM is sent after a 200 OK INVITE is received. PSTN XML HighLayerCompatibility mapped into the ATP in the ANM* Ensure that the SUT, having sent the ACM message, on receipt of a 200 OK Message containing the **PSTN XML HighLayerCompatibility** set to HLC_VALUE1 • sends the ANM message with the **ATP** including the **HLC** set to HLC_VALUE1.	
SIP Parameter values:	200 OK; **PSTN XML HighLayerCompatibility**: HLC_VALUE2 (PIXIT)	
ISUP Parameter values:	IAM; **ATP HLC1**: HLC_VALUE1 (PIXIT) **ATP HLC2**: HLC_VALUE2 (PIXIT) ANM; **ATP HLC**: HLC_VALUE2 (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP305005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/	
SIP selection criteria:	PICS 4/18 AND PICS 2/3	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*An ANM is sent after a 200 OK INVITE is received. PSTN XML BearerCompatibility mapped into the ATP and TMU in the ANM* Ensure that the SUT, having received the IAM message indicating BC fallback and having sent the ACM message, on receipt of a 200 OK Message with **PSTN XML BeaererCapability** set to BC_VALUE, • sends the ANM message with the **ATP** including the **BC** set to BC_VALUE and the TMU set to TMU_VALUE.	
SIP Parameter values:	INVITE; **PSTN XML first Bearer Capability**: INVITE _BC1 **PSTN XML second Bearer Capability**: INVITE _BC2 200 OK; **PSTN XML BeaererCapability**: BC_VALUE	
ISUP Parameter values:	IAM; **USI**: Speech/audio3Kbit/s, G.711 A-law **USI** prime: Unrestr. Digital info T/A, G.711 A-law **TMR**: 64 kbit/s preferred **TMR prime**: Speech/audio3Kbit/s ANM; **ATP BC**: BC_VALUE **TMU**: USI_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Values and additional selection criteria for test purposes TP TP305005		
VA_01	TMU_VALUE: speech ISUP_VALUE: UDI/TA	BC_VALUE: speech
VA_02	TMU_VALUE: 3,1 kHz ISUP_VALUE: UDI/TA	BC_VALUE: 3,1 kHz

TP305006	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/	
SIP selection criteria:	PICS 4/18 AND PICS 2/3	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*An ANM is sent after a 200 OK INVITE is received. No PSTN XML BearerCompatibility contained in the 200. Sending of TMU in the ANM* Ensure that the SUT, having received the IAM message indicating BC fallback and having sent the ACM message, on receipt of a 200 OK Message without **PSTN XML BeaererCapability**, • sends the ANM message with the **ATP** including the **BC** set to USI_VALUE and the TMU set to TMU_VALUE.	
SIP Parameter values:	INVITE; **PSTN XML first Bearer Capability**: INVITE _BC1 **PSTN XML second Bearer Capability**: INVITE _BC2 200 OK; **no BC**	
ISUP Parameter values:	IAM; **USI**: Speech/audio3Kbit/s, G.711 A-law **USI** prime: Unrestr. Digital info T/A, G.711 A-law **TMR**: 64 kbit/s preferred **TMR prime**: Speech/audio3Kbit/s ANM; **ATP BC**: USI_VALUE **TMU**: USI_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Values and additional selection criteria for test purposes TP305006	
VA_01	TMU_VALUE: speech ISUP_VALUE: UDI/TA
VA_02	TMU_VALUE: 3,1 kHz audio ISUP_VALUE: UDI/TA

TP305007	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/	
SIP selection criteria:	PICS 4/18 AND PICS 2/3	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*An ANM is sent after a 200 OK INVITE is received. PSTN XML HighLayerCompatibility and ProgressIndicator #5, mapped into the ATP in the ANM* Ensure that the SUT, having sent the ACM message, on receipt of a 200 OK Message containing the **PSTN XML HighLayerCompatibility** set to HLC_VALUE1 and the **PSTN XML ProgressIndicator** set to "Interworking has occurred and has resulted in a telecommunication service change (#5)" • sends the ANM message with the **ATP** including the **HLC** set to HLC_VALUE1 and the **Progress Indicator** set to "Interworking has occurred and has resulted in a telecommunication service change (#5)".	
SIP Parameter values:	200 OK INVITE **PSTN XML HighLayerCompatibility**: HLC_VALUE1 (PIXIT) **PSTN XML ProgressIndicator**: Interworking has occurred and has resulted in a telecommunication service change (#5)	
ISUP Parameter values:	IAM; **ATP HLC1**: HLC_VALUE1 (PIXIT) **ATP HLC2**: HLC_VALUE2 (PIXIT) ANM; **ATP HLC**: HLC_VALUE1 (PIXIT) **ATP Progress Indicator**: Interworking has occurred and has resulted in a telecommunication service change (#5)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP305008	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/	
SIP selection criteria:	PICS 4/18 AND PICS 2/3	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*An ANM is sent after a 200 OK INVITE is received. PSTN XML BearerCapability and ProgressIndicator #5, mapped into the ATP and TMU in the ANM* Ensure that the SUT, having received the IAM message indicating BC fallback and having sent the ACM message, on receipt of a 200 OK Message with **PSTN XML BearerCapability** information element set to BC_VALUE and the **PSTN XML ProgressIndicator** set to "Interworking has occurred and has resulted in a telecommunication service change (#5)" • sends the ANM message with the **ATP** including the **BC** set to BC_VALUE and the **TMU** set to USI_VALUE and the **Progress Indicator** set to "Interworking has occurred and has resulted in a telecommunication service change (#5)".	
SIP Parameter values:	INVITE: **PSTN XML first Bearer Capability**: INVITE _BC1 **PSTN XML second Bearer Capability**: INVITE _BC2 200 OK: **PSTN XML BearerCapability**: BC_VALUE **PSTN XML ProgressIndicator**: Interworking has occurred and has resulted in a telecommunication service change (#5)	
ISUP Parameter values:	IAM; **USI**: Speech/audio3Kbit/s, G.711 A-law **USI** prime: Unrestr. Digital info T/A, G.711 A-law **TMR**: 64 kbit/s preferred **TMR prime**: Speech/audio3Kbit/s ANM; **ATP BC**: BC_VALUE **ATP Progress Indicator**: Interworking has occurred and has resulted in a telecommunication service change (#5) **TMU**: TMU_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Values and additional selection criteria for test purposes TP305008		
VA_01	TMU_VALUE: speech ISUP_VALUE: UDI/TA	BC_VALUE: speech
VA_02	TMU_VALUE: 3,1 kHz ISUP_VALUE: UDI/TA	BC_VALUE: 3,1 kHz

6.2.2.6 Sending of the CON message

TP306001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	*CON is sent after 200 was received* Ensure that the SUT, having not sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running): • Send CON as determined by BICC/ISUP procedures. • Stop any existing awaiting answer indication (e.g. ringing tone) BCI encoded as followed: **Interworking indicator**: interworking encountered **ISUP indicator**: ISUP not used all the way **ISDN access indicator**: terminating access non-ISDN	
SIP Parameter values:	200 OK INVITE;	
ISUP Parameter values:	CON: **Interworking indicator**: interworking encountered **ISUP indicator**: ISUP not used all the way **ISDN access indicator**: terminating access non-ISDN	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE CON ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP306002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 4/9 AND PICS 4/17	
Test purpose:	*IAM received with TMR 64 kBit/s. BCI in the CON indicates ISDN access* Ensure that the SUT, having not sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running): • Send CON as determined by BICC/ISUP procedures. • Stop any existing awaiting answer indication (e.g. ringing tone) BCI encoded as follows: **interworking indicator**: no interworking encountered (0) **ISUP indicator**: ISUP used all the way **ISDN access indicator:** "terminating access ISDN"	
SIP Parameter values:	200 OK INVITE	
ISUP Parameter values:	CON: **interworking indicator**: no interworking encountered (0) **ISUP indicator**: ISUP used all the way **ISDN access indicator:** "terminating access ISDN"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE CON ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP306003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*A CON is sent after a 200 OK INVITE is received. PSTN XML ProgressIndicator #x mapped into the ATP in the CON* Ensure that on receipt of a 200 OK message containing the **PSTN XML ProgressIndicator** set to PI_VALUE sends the CON message with the **ATP** including the **PSTN XML ProgressIndicator** set to PI_VALUE.	
SIP Parameter values:	200 OK INVITE: **PSTN XML ProgressIndicator**: PI_VALUE (PIXIT)	
ISUP Parameter values:	ANM: **ATP Progress Indicator**: PI_VALUE (PIXIT)	
Comments:	**ISUP/BICC** SUT **SIP** IAM → → INVITE CON ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP306004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/	
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
Test purpose:	*A CON is sent after a 200 OK INVITE is received. PSTN XML LowLayerCompatibility mapped into the ATP in the CON* Ensure that on receipt of a 200 OK Message containing the **PSTN XML LowLayerCompatibility** set to LLC_VALUE sends the CON message with the **ATP** including the **LLC** set to LLC_VALUE.	
SIP Parameter values:	200 OK INVITE: **PSTN XML LowLayerCompatibility**: LLC_VALUE (PIXIT)	
ISUP Parameter values:	CON: **ATP LLC**: PI_VALUE (PIXIT)	
Comments:	**ISUP/BICC** SUT **SIP** IAM → → INVITE CON ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP306005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/	
SIP selection criteria:	PICS 4/18 AND PICS 2/3	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*A CON is sent after a 200 OK INVITE is received. PSTN XML HighLayerCompatibility mapped into the ATP in the CON* Ensure that on receipt of a 200 OK Message containing the **PSTN XML HighLayerCompatibility** set to HLC_VALUE1 sends the CON message with the **ATP** including the **HLC** set to HLC_VALUE1.	
SIP Parameter values:	200 OK INVITE: **PSTN XML HighLayerCompatibility**: HLC_VALUE2 (PIXIT)	
ISUP Parameter values:	IAM; **ATP HLC1**: HLC_VALUE1 (PIXIT) **ATP HLC2**: HLC_VALUE2 (PIXIT) ACON: **ATP HLC**: HLC_VALUE2 (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE CON ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP306006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/	
SIP selection criteria:	PICS 4/18 AND PICS 2/3	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*A CON is sent after a 200 OK INVITE is received. PSTN XML BearerCompatibility mapped into the ATP and TMU in the CON* Ensure that the SUT, having received the IAM message indicating BC fallback on receipt of a 200 OK Message with **PSTN XML BeaererCapability** set to BC_VALUE, sends the CON message with the **ATP** including the **BC** set to BC_VALUE and the TMU set to TMU_VALUE.	
SIP Parameter values:	INVITE; **PSTN XML first Bearer Capability**: INVITE _BC1 **PSTN XML second Bearer Capability**: INVITE _BC2 200 OK INVITE **PSTN XML BeaererCapability**: BC_VALUE	
ISUP Parameter values:	IAM; **USI**: Speech/audio3Kbit/s, G.711 A-law **USI** prime: Unrestr. Digital info T/A, G.711 A-law **TMR**: 64 kbit/s preferred **TMR prime**: Speech/audio3Kbit/s CON: **ATP BC**: BC_VALUE **TMU**: TMU_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE CON ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Values and additional selection criteria for test purposes TP TP306006		
VA_01	TMU_VALUE: speech ISUP_VALUE: UDI/TA	BC_VALUE: speech
VA_02	TMU_VALUE: 3,1 kHz ISUP_VALUE: UDI/TA	BC_VALUE: 3,1 kHz

TP306007	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/	
SIP selection criteria:	PICS 4/18 AND PICS 2/3	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*A CON is sent after a 200 OK INVITE is received. No PSTN XML BearerCompatibility contained in the 200. Sending of TMU in the CON* Ensure that the SUT, having received the IAM message indicating BC fallback on receipt of a 200 OK Message without **PSTN XML BeaererCapability**, sends the CON message with the **ATP** including the **BC** set to USI_VALUE and the TMU set to TMU_VALUE.	
SIP Parameter values:	INVITE; **PSTN XML first Bearer Capability**: INVITE _BC1 **PSTN XML second Bearer Capability**: INVITE _BC2 200 OK INVITE: **no BC**	
ISUP Parameter values:	IAM; **USI**: Speech/audio3Kbit/s, G.711 A-law **USI** prime: Unrestr. Digital info T/A, G.711 A-law **TMR**: 64 kbit/s preferred **TMR prime**: Speech/audio3Kbit/s CON: **ATP BC**: USI_VALUE **TMU**: TMU_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE CON ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Values and additional selection criteria for test purposes TP306007	
VA_01	TMU_VALUE: speech ISUP_VALUE: UDI/TA
VA_02	TMU_VALUE: 3,1 kHz audio ISUP_VALUE: UDI/TA

TP306008	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.5**
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/	
SIP selection criteria:	PICS 4/18 AND PICS 2/3	
ISUP selection criteria:	PICS 4/19	
Test purpose:	*A CON is sent after a 200 OK INVITE is received. PSTN XML HighLayerCompatibility and ProgressIndicator #5, mapped into the ATP in the CON* Ensure that the SUT on receipt of a 200 OK Message containing the **PSTN XML HighLayerCompatibility** set to HLC_VALUE1 and the **PSTN XML ProgressIndicator** set to "Interworking has occurred and has resulted in a telecommunication service change (#5)" sends the CON message with the **ATP** including the **HLC** set to HLC_VALUE1 and the **Progress Indicator** set to "Interworking has occurred and has resulted in a telecommunication service change (#5)".	
SIP Parameter values:	200 OK INVITE **PSTN XML HighLayerCompatibility**: HLC_VALUE1 (PIXIT) **PSTN XML ProgressIndicator**: Interworking has occurred and has resulted in a telecommunication service change (#5)	
ISUP Parameter values:	IAM; **ATP HLC1**: HLC_VALUE1 (PIXIT) **ATP HLC2**: HLC_VALUE2 (PIXIT) CON: **ATP HLC**: HLC_VALUE1 (PIXIT) **ATP Progress Indicator**: Interworking has occurred and has resulted in a telecommunication service change (#5)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE CON ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP306009</th><th colspan="4">SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.5</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 4/18 AND PICS 2/3</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5">PICS 4/19</td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>A CON is sent after a 200 OK INVITE is received. PSTN XML BearerCapability and ProgressIndicator #5, mapped into the ATP and TMU in the CON</i>

Ensure that the SUT, having received the IAM message indicating BC fallback and having sent the ACM message, on receipt of a 200 OK Message with PSTN XML BearerCapability information element set to BC_VALUE and the PSTN XML ProgressIndicator set to "Interworking has occurred and has resulted in a telecommunication service change (#5)"

sends the CON message with the ATP including the BC set to BC_VALUE and the TMU set to TMU_VALUE and the Progress Indicator set to "Interworking has occurred and has resulted in a telecommunication service change (#5)".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">INVITE;
PSTN XML first Bearer Capability: INVITE _BC1
PSTN XML second Bearer Capability: INVITE _BC2

200 OK INVITE
PSTN XML BearerCapability: BC_VALUE
PSTN XML ProgressIndicator: Interworking has occurred and has resulted in a telecommunication service change (#5)</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5">IAM; USI: Speech/audio3Kbit/s, G.711 A-law
USI prime: Unrestr. Digital info T/A, G.711 A-law
TMR: 64 kbit/s preferred
TMR prime: Speech/audio3Kbit/s

CON: ATP BC: BC_VALUE
ATP Progress Indicator: Interworking has occurred and has resulted in a telecommunication service change (#5)
TMU: TMU_VALUE</td></tr>
<tr><td>Comments:</td><td>ISUP/BICC</td><td></td><td>SUT</td><td></td><td>SIP</td></tr>
<tr><td></td><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td>CON</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td>REL</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td></td><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>

Values and additional selection criteria for test purposes TP306009		
VA_01	TMU_VALUE: speech ISUP_VALUE: UDI/TA	BC_VALUE: speech
VA_02	TMU_VALUE: 3,1 kHz ISUP_VALUE: UDI/TA	BC_VALUE: 3,1 kHz

6.2.2.7 Receipt of the Release message (REL)

TP307001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*REL received before INVITE was sent* Ensure that the SUT after receiving the IAM but before an INVITE has been sent. On receipt of a REL message: • no action is required on the SIP side other than to terminate local procedures if any are in progress.	
SIP Parameter values:		
ISUP Parameter values:	REL: **cause value:** CV_ISUP (PIXIT) **location**: LOC_ISUP (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → REL → RLC ←	

TP307002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*REL received, BYE is sent after ACK for 200 OK was sent before early dialogue* Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message on receipt REL message **before** a 200 OK response (any) message has been received which establishes a confirmed dialogue: • The SUT shall hold the REL message until a SIP 200 OK INVITE response has been received. • The SUT shall send a BYE request. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP.	
SIP Parameter values:	**cause value:** CV_SIP (PIXIT)	
ISUP Parameter values:	REL: **cause value:** CV_ISUP (PIXIT) **location**: LOC_ISUP (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE REL → RLC ← ← 200 OK INVITE → ACK → BYE ← 200 OK BYE	

TP307003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*200 OK INVITE received before 200 OK CANCEL was received. A BYE is sent* Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message **before** a 200 OK response message has been received: • The SUT shall hold the REL message. A **CANCEL** is sent when any SIP response was been received. • On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a **BYE** request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP.	
SIP Parameter values:	BYE: **cause value:** CV_SIP (PIXIT)	
ISUP Parameter values:	REL: **cause value:** CV_ISUP (PIXIT) **location**: LOC_ISUP (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 100 Trying REL → RLC ← → CANCEL ← 200 OK INVITE ← 200 OK CANCEL → ACK → BYE ← 200 OK BYE	

TP307004	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*REL received before early dialogue is established.* Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message **before** an early dialogue with the message defined as SIP_MESSAGE has been established: • The SUT shall hold the REL message until a **SIP_MESSAGE_VA** response has been received. • The SUT shall send a **CANCEL** request. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP.	
SIP Parameter values:	CANCEL: **cause value:** CV_SIP (PIXIT)	
ISUP Parameter values:	REL: **cause value:** CV_ISUP (PIXIT) **location**: LOC_ISUP (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE REL → RLC ← ← SIP_MESSAGE_VA → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK	

TP307005	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*REL received, BYE is sent after ACK for 200 OK was sent in early dialogue* Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message **after** a 200 OK response message has been received: • The SUT shall send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP.	
SIP Parameter values:	BYE: **cause value:** CV_SIP (PIXIT)	
ISUP Parameter values:	REL: **cause value:** CV_ISUP (PIXIT) **location**: LOC_ISUP (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE REL → RLC ← → ACK → BYE ← 200 OK BYE	

TP307006	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.8**
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*REL received, BYE is sent after ACK for 200 OK was sent in early dialogue established by several messages* Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message **after** an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: • The SUT shall send a CANCEL request. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP.	
SIP Parameter values:	CANCEL: **cause value:** CV_SIP (PIXIT)	
ISUP Parameter values:	REL: **cause value:** CV_ISUP (PIXIT) **location**: LOC_ISUP (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← SIP_MESSAGE_VA REL → RLC ← → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK	

Table 11

Values for test purpose TP307004; TP307006	
VA	**SIP MESSAGE_VA**
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

Table 12

Values for test purposes 307004 - 307006		
←SIP Message **Reason header field** **CV_SIP**		**← REL** **Cause Indicators parameter** **CV_ISUP**
VA_1	Normal call clearing # 16	Normal call clearing # 16
VA_2	Normal, unspecified # 31	Normal, unspecified # 31
VA_3	Temporary failure # 41	Temporary failure # 41
VA_4	Invalid message, unspecified # 95	Invalid message, unspecified # 95
VA_5	Recovery on timer expiry # 102	Recovery on timer expiry # 102
VA_6	Protocol error, unspecified # 111	Protocol error, unspecified # 111

Table 13: Mapping of Cause Indicators parameter into SIP Reason header fields

Cause indications parameter field	Value of parameter field	component of SIP Reason header field	Component value
-	-	Protocol	"*ITU-T Rec Q.850 [*5*]*"
Cause Value	"*XX*" (see note 1)	Protocol-cause	"*cause= XX*" (see note 1)
-	-	Reason-text	Should be filled with the definition text as stated in ITU-T Recommendation Q.850 [5] (see note 2)
NOTE 1: "XX" is the Cause Value as defined in ITU-T Rec Q.850 [5]. NOTE 2: Due to the fact that the Cause Indications parameter does not include the definition text as defined in table1/ITU-T Recommendation Q.850 [5] this is based on provisioning in the O-IWU.			

6.2.2.8 Sending of a REL message / receipt of a backward BYE

TP308001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.7**
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*BYE received, REL cause #16 is sent* Ensure that the SUT after receiving the IAM sends out an INVITE message and on receipt of a BYE message where a Reason header field with ITU-T Recommendation Q.850 [5] Cause Value is **not** included: • sends a REL message with the **Cause value** Value No. 16 ("*normal clearing*").	
SIP Parameter values:		
ISUP Parameter values:	REL; Cause value "Normal call clearing"	
Comments:	**ISUP/BICC SUT SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL ← ← BYE RLC → → 200 OK BYE	

TP308002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.7**
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	PICS 4/11	
ISUP selection criteria:		
Test purpose:	*BYE Reason header #x received, REL cause #x is sent* Ensure that the SUT after receiving the IAM sends out a INVITE message and on receipt of a BYE message where a Reason header field with ITU-T Recommendation Q.850 [5] Cause Value is included: • sends a REL message. The Cause Value is in the Reason header filed mapped to the ISUP Cause Value field in the ISUP REL.	
SIP Parameter values:	BYE **cause value:** CV_SIP (PIXIT)	
ISUP Parameter values:	REL; **cause value:** CV_ISUP (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL ← ← BYE RLC → → 200 OK BYE	

Table 14: Mapping of SIP Reason header fields into Cause Indicators parameter

component of SIP Reason header field	Component value	BICC/ISUP Parameter / field	value
Protocol	"*ITU-T Rec Q.850 [5]*"	Cause Indication parameter	-
protocol-cause	"*cause = XX*" (see note)	Cause Value	"*XX*" (see note)
-	-	Location	*"network beyond interworking point"*
NOTE: "XX" is the Cause Value as defined in ITU-T Recommendation Q.850 [5].			

TP308003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.7**
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Final response without Reason header received, mapping in REL* Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is **not** included defined as SIP_Failure_VA: • sends a REL message with the Cause value set to CV_ISUP.	
SIP Parameter values:		
ISUP Parameter values:	REL; **cause value:** CV_ISUP	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE REL ← ← SIP_Failure_VA RLC → → ACK	

Table 15

Values for test purpose TP308003		
VA	**←REL (Cause Value) CV_ ISUP**	**←4XX/5XX/6XX SIP message SIP_Failure_VA**
VA_01	127 Interworking	400 Bad Request
VA_02	127 Interworking	401 Unauthorised
VA_03	127 Interworking	402 Payment Required
VA_04	127 Interworking	403 Forbidden
VA_05	127 Interworking	405 Method Not Allowed
VA_06	127 Interworking	406 Not Acceptable
VA_07	127 Interworking	407 Proxy authentication required
VA_08	127 Interworking	408 Request Timeout
VA_09	22 Number changed (without diagnostic)	410 Gone
VA_10	127 Interworking	413 Request Entity too long
VA_11	127 Interworking	414 Request-uri too long
VA_12	127 Interworking	415 Unsupported Media type
VA_13	127 Interworking	416 Unsupported URI scheme
VA_14	127 Interworking	420 Bad Extension
VA_15	127 Interworking	421 Extension required
VA_16	127 Interworking	423 Interval Too Brief
VA_17	20 Subscriber absent	480 Temporarily Unavailable
VA_18	127 Interworking	481 Call/Transaction does not exist
VA_19	127 Interworking	482 Loop Detected
VA_20	127 Interworking	483 Too many hops
VA_21	127 Interworking	485 Ambiguous
VA_22	17 User busy	486 Busy Here
VA_23	127 Interworking	488 Not acceptable here
VA_24	127 Interworking	493 Undecipherable
VA_25	127 Interworking	500 Server Internal error
VA_26	127 Interworking	501 Not implemented
VA_27	127 Interworking	502 Bad Gateway
VA_28	127 Interworking	503 Service Unavailable
VA_29	127 Interworking	504 Server timeout
VA_30	127 Interworking	505 Version not supported
VA_31	127 Interworking	513 Message too large
VA_32	127 Interworking	580 Precondition failure
VA_33	17 User busy	600 Busy Everywhere
VA_34	21 Call rejected	603 Decline
VA_35	1 Unallocated number	604 Does not exist anywhere
VA_36	127 Interworking	606 Not acceptable

TP308004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.7**
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Final response without Reason header received, mapping in REL* Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with ITU-T Recommendation Q.850 [5] Cause Value CV_ISUP is included defined as SIP_Failure_VA: • sends a REL message with the Cause value set to CV_ISUP.	
SIP Parameter values:		
ISUP Parameter values:	REL; **cause value:** CV_ISUP	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE REL(#xx) ← ← SIP_Failure_VA RLC → → ACK	

Table 16

Values for test purpose TP308004, TP308005		
VA	**←REL (Cause Value PIXIT)**	**←4XX/5XX/6XX SIP message SIP_Failure_VA**
VA_01	CV_ ISUP	400 Bad Request
VA_02	CV_ ISUP	401 Unauthorised
VA_03	CV_ ISUP	402 Payment Required
VA_04	CV_ ISUP	403 Forbidden
VA_05	CV_ ISUP	405 Method Not Allowed
VA_06	CV_ ISUP	406 Not Acceptable
VA_07	CV_ ISUP	407 Proxy authentication required
VA_08	CV_ ISUP	408 Request Timeout
VA_09	CV_ ISUP	410 Gone
VA_10	CV_ ISUP	413 Request Entity too long
VA_11	CV_ ISUP	414 Request-uri too long
VA_12	CV_ ISUP	415 Unsupported Media type
VA_13	CV_ ISUP	416 Unsupported URI scheme
VA_14	CV_ ISUP	420 Bad Extension
VA_15	CV_ ISUP	421 Extension required
VA_16	CV_ ISUP	423 Interval Too Brief
VA_17	CV_ ISUP	480 Temporarily Unavailable
VA_18	CV_ ISUP	481 Call/Transaction does not exist
VA_19	CV_ ISUP	482 Loop Detected
VA_20	CV_ ISUP	483 Too many hops
VA_21	CV_ ISUP	485 Ambiguous
VA_22	CV_ ISUP	486 Busy Here
VA_23	CV_ ISUP	488 Not acceptable here
VA_24	CV_ ISUP	493 Undecipherable
VA_25	CV_ ISUP	500 Server Internal error
VA_26	CV_ ISUP	501 Not implemented
VA_27	CV_ ISUP	502 Bad Gateway
VA_28	CV_ ISUP	503 Service Unavailable
VA_29	CV_ ISUP	504 Server timeout
VA_30	CV_ ISUP	505 Version not supported
VA_31	CV_ ISUP	513 Message too large
VA_32	CV_ ISUP	580 Precondition failure
VA_33	CV_ ISUP	600 Busy Everywhere
VA_34	CV_ ISUP	603 Decline
VA_35	CV_ ISUP	604 Does not exist anywhere
VA_36	CV_ ISUP	606 Not acceptable

TP308005	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.7**
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Final response contains a Reason header in early dialogue received* Ensure that the SUT after receiving the IAM sends out an INVITE message, a SIP message defined as **SIP_MESSAGE_VA** has been received on receipt of a Failure message (4xx, 5xx, 6xx) defined as **SIP_Failure_VA** where a Reason header field with ITU-T Recommendation Q.850 [5] Cause Value is included: • sends a REL message. The Cause Value in the header field set to CV_SIP is mapped to the ISUP Cause Value field in the ISUP REL message with the Cause value set to CV_ ISUP.	
SIP Parameter values:	CV_SIP (PIXIT)	
ISUP Parameter values:	CV_ ISUP (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← SIP MESSAGE_VA REL ← ← SIP_Failure_VA RLC → → ACK	

Table 17

Values for test purpose TP308005	
VA	**SIP MESSAGE_VA**
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

TP308006	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.7**
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Final response without Reason header received, mapping in REL* Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a response message (3xx) defined as **SIP_Response_VA,** the SUT: • sends a REL message with the **Cause value** 127 Interworking.	
SIP Parameter values:		
ISUP Parameter values:	REL; **cause value:** 127	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE REL(#127) ← ← SIP_Response_VA RLC → → ACK	

Table 18

Values for test purposes TP308006		
VA	**←REL (Cause Value)** **CV_ ISUP**	**←3XX SIP message** **SIP_Response_VA**
VA_01	127 Interworking	300 Multiple Choices
VA_02	127 Interworking	301 Moved Permanently
VA_03	127 Interworking	302 Move Temporarily
VA_04	127 Interworking	305 Use Proxy
VA_05	127 Interworking	380 Alternative Service

6.2.2.9 Autonomous release at O-MGCF

TP309001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.12.1**
TSS reference:	ISUP-SIP/Basic call/Autonomous release/	
SIP selection criteria:	PICS 3/2	
ISUP selection criteria:		
Test purpose:	*Overlap supported, REL is sent when 404/484 received and Ti/w3 is expired* Ensure that the SUT a On receipt of a 484 Address Incomplete or 404 Not Found response for the current INVITE (i.e. there are no other pending INVITE transactions for this call), if the SUT is configured to propagate overlap signalling into the SIP network, the SUT: • Shall not send a REL message immediately and shall instead start timer TOIW3. The REL message shall only be sent if TI/W3 expires. • The REL message containes the Cause Value 28.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE **CASE A** ← 484 Address Incomplete → ACK Start timer $T_{I/W3}$ Timeout $T_{I/W3}$ REL #28 ← RLC → **CASE B** ← 404 Not Found → ACK Start timer $T_{I/W3}$ Timeout $T_{I/W3}$ REL #28 ← RLC →	

TP309002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.12.1**
TSS reference:	ISUP-SIP/Basic call/Autonomous release/	
SIP selection criteria:	NOT PICS 3/2	
ISUP selection criteria:		
Test purpose:	*Overlap not supported, REL is sent when 404/484 received* Ensure that the SUT on receipt of a 484 Address Incomplete response for the current INVITE (i.e. there are no other pending INVITE transactions for this call), if the O-MGCF is not configured to propagate overlap signalling into the SIP network then the timer shall not be started and the: • REL shall be sent immediately to the BICC/ISUP network.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE REL #28 ← ← 484 Address Incomplete RLC → → ACK	

TP309003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.18**
TSS reference:	ISUP-SIP/Basic call/Autonomous release/	
SIP selection criteria:	PICS 4/5 AND PICS 4/11	
ISUP selection criteria:	PICS 4/2	
Test purpose:	*Preconditions supported, call setup released when COT(failed) received* Ensure that the SUT a on receipt of a COT "failed" and preconditions used, the SUT: • sends a CANCEL to the SIP network.	
SIP Parameter values:		
ISUP Parameter values:	IAM: Nature of connection indicators "continuity check required on this circuit"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE COT(failed) → → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK	

TP309004	**SIP reference: RFC 3261 [6]**	**ISUP reference: Es 283 027 [1], clause 7.7.3**
TSS reference:	ISUP-SIP/Basic call/Autonomous release/	
SIP selection criteria:	PICS 4/5 AND PICS 4/11	
ISUP selection criteria:	PICS 4/2	
Test purpose:	*Preconditions supported, call setup released when T8 expired* Ensure that the SUT when the ISUP/BICC timer T8 is expired and preconditions used, the SUT: • sends a CANCEL or BYE to the SIP network.	
SIP Parameter values:		
ISUP Parameter values:	IAM: Nature of connection indicators "continuity check required on this circuit"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE T8 expirees → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK	

TP309005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.16**
TSS reference:	ISUP-SIP/Basic call/Autonomous release/	
SIP selection criteria:	PICS 4/7 AND PICS 4/15	
ISUP selection criteria:	PICS 4/2	
Test purpose:	*Preconditions supported, 580 mapped in REL #47* Ensure that the SUT when the resource reservation is unsuccessful and preconditions used, the SUT responds to an INVITE: • send a REL with cause value # 47	
SIP Parameter values:		
ISUP Parameter values:	IAM: Nature of connection indicators "continuity check required on this circuit"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE REL ← ← 580 Precondition Failure RLC → → ACK	

TP309006	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.17.2**
TSS reference:	ISUP-SIP/Basic call/Autonomous release/	
SIP selection criteria:	PICS 4/7 AND PICS 4/15	
ISUP selection criteria:	PICS 4/2	
Test purpose:	*Preconditions supported, 580 mapped in REL #47* Ensure that the SUT when the resource reservation is unsuccessful and preconditions used, the SUT responds to an UPDATE: • send a REL with cause value # 47	
SIP Parameter values:		
ISUP Parameter values:	IAM: Nature of connection indicators "continuity check required on this circuit"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 183 Session Progress → PRACK ← 200 OK PRACK REL ← ← 580 Precondition Failure RLC → → ACK	

6.2.2.10 Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented

6.2.2.10.1 Receipt of Reset Circuit message (RSC)

TP310001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC)	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*RSC received while an INVITE was not sent* Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of a RSC message: • no action is required on the SIP side other than to terminate local procedures if any are in progress.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → RSC → RLC ←	

<table>
<tr><th>TP310002</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.2.15</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC)</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>RSC received while no response for an INVITE is received</i>
Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message before a SIP MESSAGE_VA response message has been received:

• The SUT shall hold the RSC message until a SIP response has been received.
• The SUT shall send a CANCEL request.
• A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>Comments:</td><td colspan="2">ISUP/BICC SUT SIP
IAM → → INVITE
RSC →
RLC ← ← SIP MESSAGE_VA
→ CANCEL
← 200 OK CANCEL
← 487 Request terminated
→ ACK</td></tr>
</table>

Table 19

<table>
<tr><th colspan="2">Values for test purpose TP310002</th></tr>
<tr><th>VA</th><th>SIP MESSAGE_VA</th></tr>
<tr><td>VA_1</td><td>100 Trying</td></tr>
<tr><td>VA_2</td><td>180 Ringing</td></tr>
<tr><td>VA_3</td><td>181 Call Is Being Forwarded</td></tr>
<tr><td>VA_4</td><td>182 Queued</td></tr>
<tr><td>VA_5</td><td>183 Session Progress</td></tr>
</table>

TP310003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC)	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*RSC received. While CANCEL is sent, a 200 OK INVITE is received* Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message **before** a 200 OK response message has been received: • On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. • A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 100 Trying RSC → → CANCEL RLC ← ← 200 OK INVITE → ACK ← 200 OK CANCEL → BYE ← 200 OK BYE	

TP310004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC)	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*RSC received after the ACK for a 200 OK INVITE was sent. A BYE is sent* Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a BYE message on receipt RSC message **after** a 200 OK response message has been received: • The SUT shall send a BYE request. • A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK RSC → → BYE RLC ← ← 200 OK BYE	

TP310005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC)	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*RSC in early dialogue received. A CANCEL is sent* Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message **after** an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: • The SUT shall send a CANCEL or BYE request. • A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← SIP MESSAGE_VA RSC → RLC ← → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK	

Table 20

Values for test purpose; TP310005	
VA	**SIP MESSAGE_VA**
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

6.2.2.10.2 Receipt of Circuit group reset message (GRS)

TP311001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group reset message (GRS)	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*GRS received while an INVITE was not sent* Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of GRS message: • no action is required on the SIP side other than to terminate local procedures if any are in progress.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → GRS → GRA ←	

TP311002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group reset message (GRS)	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*GRS received while no response for an INVITE is received* Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message **before SIP MESSAGE_VA** response message has been received: • The SUT shall hold the GRS message until a SIP response has been received. • The SUT shall send a CANCEL request. • A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE GRS → GRA ← ← SIP MESSAGE_VA → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK	

Table 21

Values for test purpose TP311002	
VA	**SIP MESSAGE_VA**
VA_1	100 Trying
VA_2	180 Ringing
VA_3	181 Call Is Being Forwarded
VA_4	182 Queued
VA_5	183 Session Progress

TP311003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group reset message (GRS)	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*GRS received. While CANCEL is sent, a 200 OK INVITE is received* Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message **before** a 200 OK response message has been received: • The SUT shall hold the GRS message until a response has been received. A CANCEL is sent. • On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 100 Trying GRS → → CANCEL GRA ← ← 200 OK INVITE → ACK ← 200 OK CANCEL → BYE ← 200 OK BYE	

TP311004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group reset message (GRS)	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*GRS received after the ACK for a 200 OK INVITE was sent. A BYE is sent* Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a BYE message on receipt GRS message **after** a 200 OK response message has been received: • The SUT shall send a BYE request. • A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is isadded to the SIP message to be sent by the SIP side of the O-MGCF.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK GRS → → BYE GRA ← ← 200 OK BYE	

TP311005	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group reset message (GRS)	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*GRS in early dialogue received. A CANCEL is sent* Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message **after** an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: • The SUT shall send a CANCEL request. • A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← SIP MESSAGE_VA GRS → GRA ← → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK	

Table 22

Values for test purpose TP309009; TP311005	
VA	**SIP MESSAGE_VA**
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

TP311006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group reset message (GRS)	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*GRS for more than one CIC received. Send a BYE for each circuit* Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a GRS message were the Range Parameter value is bigger than *"1"*: • the SUT shall send a BYE requests for each call association.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM(1) → → INVITE(1) ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK IAM(2) → → INVITE(2) ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK GRS(1) → → BYE(1) GRA ← ← 200 OK BYE → BYE(2) ← 200 OK BYE	
NOTE: BYE(1) and BYTE(2) possible received in reverse order.		

6.2.2.10.3 Receipt of Circuit group blocking message (CGB)

TP312001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CGB received while an INVITE was not sent* Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented": • no action is required on the SIP side other than to terminate local procedures if any are in progress.	
SIP Parameter values:		
ISUP Parameter values:	CGB/CGBA: Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → CBG → CGBA ←	

TP312002	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CGB received while no response for an INVITE is received* Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **before** a **SIP MESSAGE_VA** response message has been received: • The SUT shall hold the CGB message until a SIP 200 OK response has been received. • The SUT shall send a CANCEL request. • A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.	
SIP Parameter values:		
ISUP Parameter values:	CGB/CGBA: Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE CGB → CGBA ← ← SIP MESSAGE_VA → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK	

Table 23

Values for test purpose TP312002	
VA	**SIP MESSAGE_VA**
VA_1	100 Trying
VA_2	180 Ringing
VA_3	181 Call Is Being Forwarded
VA_4	182 Queued
VA_5	183 Session Progress

TP312003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CGB received. While CANCEL is sent, a 200 OK INVITE is received* Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **before** a 200 OK response message has been received: • On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent • A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.	
SIP Parameter values:		
ISUP Parameter values:	CGB/CGBA: Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← 100 Trying CGB → → CANCEL CGBA ← ← 200 OK INVITE → ACK ← 200 OK CANCEL → BYE ← 200 OK BYE	

TP312004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CGB received after the ACK for a 200 OK INVITE was sent. A BYE is sent* Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a BYE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **after** a 200 OK response message has been received: • The SUT shall send a BYE request. • A Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.	
SIP Parameter values:		
ISUP Parameter values:	CGB/CGBA: Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK CGB → → BYE CGBA ← ← 200 OK BYE	

TP312005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CGB in early dialogue received. A CANCEL is sent* Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **after** an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: • The SUT shall send a CANCEL request. • A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.	
SIP Parameter values:		
ISUP Parameter values:	CGB/CGBA: Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ← SIP MESSAGE_VA CGB → CGBA ← → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK	

Table 24

Values for test purpose ; TP312005	
VA	**SIP MESSAGE_VA**
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

TP312006	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.2.15**
TSS reference:	ISUP-SIP/Basic call/ Receipt of Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CGB for more than one CIC received. Send a BYE for each circuit* Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" **where the Range and Status Parameter value is bigger than *"1"***: • the SUT shall send a BYE requests for each call association.	
SIP Parameter values:		
ISUP Parameter values:	CGB/CGBA: Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented"	
omments:	**ISUP/BICC** **SUT** **SIP** IAM(1) → → INVITE(1) ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK IAM(2) → → INVITE(2) ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK CGB(1) → → BYE(1) CGBA ← ← 200 OK BYE → BYE(2) ← 200 OK BYE	
NOTE: BYE(1) and BYTE(2) possible received in reverse order.		

6.3.1 Interworking from SIP to ISUP (Incoming Call)

6.3.1.1 Calling Line Identification (CLI)

TP501001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*P-Asserted-Identity not in E.164 format and From header not in the E.164 Format, no Privacy header. Send Calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received;** • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • a Privacy header field **has not been received**. sends an **IAM** message with the **Calling party number parameter** coded: Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 25: Values for test purposes TP501001

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*P-Asserted-Identity not in E.164 format and From header not in the E.164 Format, Privacy value none. Send Calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: - the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; - the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; - a Privacy header field was received and the priv-value component is set to "none". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP501003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*P-Asserted-Identity not in E.164 format and From header not in the E.164 Format, Privacy value header. Send Calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • a Privacy header field was received and the priv-value component is set to "header". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000	
SIP Parameter values:		
ISUP Parameter values:		
Comments:		

SIP		**SUT**		**ISUP**
INVITE	→		→	IAM
180 Ringing	←		←	ACM
		Ringing tone		
200 OK INVITE	←		←	ANM
ACK	→			
		Conversation		
BYE	→		→	REL
200 OK BYE	←		←	RLC

<table>
<tr><td>TP501004</td><td>SIP reference: RFC 3261 [6]</td><td colspan="4">ISUP reference:
ES 283 027 [1], clause 7.2.3.1.2.6</td></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISUP/SS/CLI/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>P-Asserted-Identity not in E.164 format and From header not in the E.164 Format, Privacy value user. Send Calling party number</i>
Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received;
• the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received;
• a Privacy header field was received and the priv-value component is set to "user".
sends an IAM message with the Calling party number parameter coded:
Address signals = absent
Screening indicator = network provided
Number Incomplete Indicator = incomplete
Numbering plan indicator = 000
Address Presentation Restricted Indicator = PIXIT
Nature of address indicator = 0000000</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5"></td></tr>
<tr><td>Comments:</td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td></td><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td></td><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td></td><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td></td><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td></td><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP501005	**SIP reference: RFC 3261 [6]**			**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.2.6**	
TSS reference:	SIP-ISUP/SS/CLI/				
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	*P-Asserted-Identity not in E.164 format and From header not in the E.164 Format, Privacy value id. Send Calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • a Privacy header field was received and the priv-value component is set to "id". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000				
SIP Parameter values:					
ISUP Parameter values:					
Comments:	**SIP**		**SUT**		**ISUP**
	INVITE	→		→	IAM
	180 Ringing	←		←	ACM
			Ringing tone		
	200 OK INVITE	←		←	ANM
	ACK	→			
			Conversation		
	BYE	→		→	REL
	200 OK BYE	←		←	RLC

TP501006	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	*P-Asserted-Identity not in E.164 format and From header in the E.164 Format, no Privacy header received. Send Calling party number and Additional calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • a Privacy header field **has not been received**. sends an **IAM** message with the **Calling party number parameter** coded: Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000 with the **Generic number parameter** coded: Address signals = derrived from the From header Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 26: Values for test purposes TP501006

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501007	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	*P-Asserted-Identity not in E.164 format and From header in the E.164 Format, Privacy value none received. Send Calling party number and Additional calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • a Privacy header field was received and the priv-value component is set to "none". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000 with the **Generic number parameter** coded: Address signals = number provided by the user Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 27: Values for test purposes TP501007

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501008	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	*P-Asserted-Identity not in E.164 format and From header in the E.164 Format, Privacy value header received. Send Calling party number and Additional calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • a Privacy header field was received and the priv-value component is set to "header". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000 with the **Generic number parameter** coded: Address signals = number provided by the user Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 28: Values for test purposes TP501008

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501009	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	*P-Asserted-Identity not in E.164 format and From header in the E.164 Format, Privacy value user received. Send Calling party number and Additional calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • a Privacy header field was received and the priv-value component is set to "user". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000 with the **Generic number parameter** coded: Address signals = number provided by the user Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 29: Values for test purposes TP501009

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501010	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	*P-Asserted-Identity not in E.164 format and From header in the E.164 Format, Privacy value id received. Send Calling party number and Additional calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • a Privacy header field was received and the priv-value component is set to "id". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator = PIXIT Nature of address indicator = 0000000 with the **Generic number parameter** coded: Address signals = number provided by the user Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 30: Values for test purposes TP501010

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501011	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*P-Asserted-Identity in E.164 format and From header not in the E.164 Format, no Privacy header received. Send Calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • a Privacy header field has **not been received**. sends an **IAM** message with the **Calling party number parameter** coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 31: Values for test purposes TP501011

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501012	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*P-Asserted-Identity in E.164 format and From header not in the E.164 Format, Privacy value none received. Send Calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • a Privacy header field was received and the priv-value component is set to "none". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 32: Values for test purposes TP501012

VA	**The next BICC/ISUP node is located**	**NoA_VALUE**	**Number parameter address format**
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501013	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*P-Asserted-Identity in E.164 format and From header not in the E.164 Format, Privacy value header received. Send Calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • a Privacy header field was received and the priv-value component is set to "header". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP501014	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*P-Asserted-Identity in E.164 format and From header not in the E.164 Format, Privacy value user received. Send Calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**; • a Privacy header field was received and the priv-value component is set to "user". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

<table>
<tr><td>**TP501015**</td><td colspan="2">**SIP reference: RFC 3261 [6]**</td><td colspan="3">**ISUP reference:
ES 283 027 [1], clause 7.2.3.1.2.6**</td></tr>
<tr><td>**TSS reference:**</td><td colspan="5">SIP-ISUP/SS/CLI/</td></tr>
<tr><td>**SIP selection criteria:**</td><td colspan="5"></td></tr>
<tr><td>**ISUP selection criteria:**</td><td colspan="5"></td></tr>
<tr><td>**Test purpose:**</td><td colspan="5">*P-Asserted-Identity in E.164 format and From header not in the E.164 Format, Privacy value id received. Send Calling party number*
Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**;
• the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been received**;
• a Privacy header field was received and the priv-value component is set to "id".
sends an **IAM** message with the **Calling party number parameter** coded:
Address signals = number derived from SIP P-Asserted-Identity
Screening indicator = network provided
Number Incomplete Indicator = complete
Numbering plan indicator = ISDN numbering plan
Address Presentation Restricted Indicator = Presentation restricted</td></tr>
<tr><td>**SIP Parameter values:**</td><td colspan="5"></td></tr>
<tr><td>**ISUP Parameter values:**</td><td colspan="5"></td></tr>
<tr><td rowspan="9">**Comments:**</td><td>**SIP**</td><td></td><td>**SUT**</td><td></td><td>**ISUP**</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP501016	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	*P-Asserted-Identity in E.164 format and From header in the E.164 Format, no Privacy header received. Send Calling party number and Additional calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • a Privacy header field has not been received. sends an **IAM** message with the **Calling party number parameter** coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE with the **Generic number parameter** coded: Address signals = number derived from the From header Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:		

SIP		**SUT**		**ISUP**
INVITE	→		→	IAM
180 Ringing	←		←	ACM
		Ringing tone		
200 OK INVITE	←		←	ANM
ACK	→			
		Conversation		
BYE	→		→	REL
200 OK BYE	←		←	RLC

Table 33: Values for test purposes TP501016

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501017	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	*P-Asserted-Identity in E.164 format and From header in the E.164 Format, Privacy value none received. Send Calling party number and Additional calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • a Privacy header field was received and the priv-value component is set to "none". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE with the **Generic number parameter** coded: Address signals = number derived from the From header Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 34: Values for test purposes TP501017

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501018	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	*P-Asserted-Identity in E.164 format and From header in the E.164 Format, Privacy value header received. Send Calling party number and Additional calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • a Privacy header field was received and the priv-value component is set to "header". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE with the **Generic number parameter** coded: Address signals = number derived from the From header Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 35: Values for test purposes TP501018

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501019	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	*P-Asserted-Identity in E.164 format and From header in the E.164 Format, Privacy value user received. Send Calling party number and Additional calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • a Privacy header field was received and the priv-value component is set to "user". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE with the **Generic number parameter** coded: Address signals = number derived from the From header Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 36: Values for test purposes TP501019

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

TP501020	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	*P-Asserted-Identity in E.164 format and From header in the E.164 Format, Privacy value id received. Send Calling party number and Additional calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has been received**; • a Privacy header field was received and the priv-value component is set to "id". sends an **IAM** message with the **Calling party number parameter** coded: Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE with the **Generic number parameter** coded: Address signals = number derived from the From header Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

<table>
<tr><th>TP501021</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
ES 283 027 [1], clause 7.2.3.1.2.6</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISUP/SS/CLI/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5">PICS 6/1</td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>P-Asserted-Identity not in E.164 format and From header not in the E.164 Format, no Privacy header. Send Calling party number network provided</i>
Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received;
• the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received;
• a Privacy header field has not been received.
sends an IAM message with the Calling party number parameter coded:
Address signals = network provided (PIXIT)
Screening indicator = network provided
Nature of address indicator = NoA_VALUE
Number Incomplete Indicator = complete
Numbering plan indicator = ISDN numbering plan
Address Presentation Restricted Indicator = PIXIT</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5"></td></tr>
<tr><td rowspan="9">Comments:</td><td>SIP</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP501022	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*P-Asserted-Identity sip URI, without user=phone and P-Asserted-Identity tel URI, no Privacy header. Send Calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP **P-Asserted-Identity** containing a SIP URI with an identity 1 in the format "+" CC+ NDC+ SN **has been received without user = phone**; • the SIP **P-Asserted-Identity** containing a Tel URI with an identity 2 in the format "+" CC+ NDC+ SN **has been received**; • a Privacy header field has not been received. sends an **IAM** message with the **Calling party number parameter** coded: Address signals = identity 2 Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

<table>
<tr><th>TP501023</th><th colspan="5">SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.2.3.1.2.6</th></tr>
<tr><td>TSS reference:</td><td colspan="6">SIP-ISUP/SS/CLI/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="6"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="6">PICS 6/1 AND PICS 6/12</td></tr>
<tr><td>Test purpose:</td><td colspan="6"><i>P-Asserted-Identity not in E.164 format, no Privacy header. Send Calling party number network provided Address not available</i>
Ensure that the SUT in the Idle state, on receipt of a INVITE message where:
• the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received;
• a Privacy header field has not been received.
sends an IAM message with the Calling party number parameter coded:
Address signals = not present
Screening indicator = network provided
Number Incomplete Indicator = incomplete
Address Presentation Restricted Indicator = <i>Address not available</i></td></tr>
<tr><td>SIP Parameter values:</td><td colspan="6"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="6"></td></tr>
<tr><td>Comments:</td><td>SIP</td><td></td><td>SUT</td><td></td><td colspan="2">ISUP</td></tr>
<tr><td></td><td>INVITE</td><td>→</td><td></td><td>→</td><td colspan="2">IAM</td></tr>
<tr><td></td><td>180 Ringing</td><td>←</td><td></td><td>←</td><td colspan="2">ACM</td></tr>
<tr><td></td><td></td><td></td><td>Ringing tone</td><td></td><td colspan="2"></td></tr>
<tr><td></td><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td colspan="2">ANM</td></tr>
<tr><td></td><td>ACK</td><td>→</td><td></td><td></td><td colspan="2"></td></tr>
<tr><td></td><td></td><td></td><td>Conversation</td><td></td><td colspan="2"></td></tr>
<tr><td></td><td>BYE</td><td>→</td><td></td><td>→</td><td colspan="2">REL</td></tr>
<tr><td></td><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td colspan="2">RLC</td></tr>
</table>

TP501024	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/1 AND PICS 6/3 AND PICS 6/12	
Test purpose:	*P-Asserted-Identity not in E.164 format and From header in the E.164 Format, no Privacy header received. Send Calling party number network provided and Additional calling party number* Ensure that the SUT in the Idle state, on receipt of a INVITE message where: • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN **has not been** received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN **has been** received; • a Privacy header field **has not been** received. sends an **IAM** message with the **Calling party number parameter** coded: Address signals = not present Screening indicator = network provided Number Incomplete Indicator = incomplete Address Presentation Restricted Indicator = *Address not available* with the Generic number parameter coded: Address signals = number derived from the From header Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 37: Values for test purposes TP501122, TP501023, TP501024

VA	The next BICC/ISUP node is located	NoA_VALUE	Number parameter address format
VA_01	in the same country	**'National (Significant) number"**	NDC+SN
VA_01	a different country	**"International number"**	CC+NDC+SN

6.3.1.2 Call Hold (HOLD)

TP502001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **7.4.10/[14]**
TSS reference:	SIP-ISUP/SS/HOLD/	
SIP selection criteria:	PICS 8/4	
ISUP selection criteria:	PICS 5/22	
Test purpose:	*Each party can hold and retrieve the remote party in the confirmed state* Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party The called party should be able to put the other party on hold The called party should be able to retrieve the other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM INVITE(sendonly) → → CPG(hold) 200 OK INVITE(recvonly) ← INVITE(sendrecv) → → CPG(retrieve) 200 OK INVITE(sendrecv) ← INVITE(sendonly) ← ← CPG(hold) 200 OK INVITE(recvonly) → INVITE(sendrecv) ← ← CPG(retrieve) 200 OK INVITE(sendrecv) →	

TP502002	**SIP reference: RFC 3261 [6]**	**ISUP reference: 7.4.10/[14]**
TSS reference:	SIP-ISUP/SS/HOLD/	
SIP selection criteria:	PICS 8/4	
ISUP selection criteria:	PICS 5/22 PICS 8/1	
Test purpose:	*The calling party can hold and retrieve the remote party in the early dialogue* Ensure that a party can put the other party on hold in the alerting state. Ensure that the party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM UPDATE(sendonly) → → CPG(hold) 200 OK UPDATE(recvonly) ← UPDATE(sendrecv) → → CPG(retrieve) 200 OK UPDATE(sendrecv) ←	

TP502003	**SIP reference: RFC 3261 [6]**	**ISUP reference: 7.4.10/[14]**
TSS reference:	SIP-ISUP/SS/HOLD/	
SIP selection criteria:	PICS 8/2	
ISUP selection criteria:	PICS 5/22	
Test purpose:	*The calling party can hold and retrieve the remote party after the calling party has provided all information to process the call* Ensure that a party can put the other party on hold after the calling user has provided all of the information necessary for processing the call. Ensure that the party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version> incremented	
ISUP Parameter values:	ACM: called party status: no indication CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval Event indicator PROGRESS (retrieve the call)	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM UPDATE(sendonly) → 200 OK UPDATE(recvonly) ← UPDATE(sendrecv) → 200 OK UPDATE(sendrecv) ←	

TP502004	**SIP reference: RFC 3261 [6]**	**ISUP reference: 7.4.10/[14]**
TSS reference:	SIP-ISUP/SS/HOLD/	
SIP selection criteria:	PICS 8/4	
ISUP selection criteria:	PICS 5/22	
Test purpose:	*A party can hold and retrieve the remote party in the confirmed state using the UPDATE method (receiving)* Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM UPDATE(sendonly) → → CPG(hold) 200 OK INVITE(recvonly) ← UPDATE(sendrecv) → → CPG(retrieve) 200 OK UPDATE(recvonly) ←	

TP502005	**SIP reference: RFC 3261 [6]**	**ISUP reference: 7.4.10/[14]**
TSS reference:	SIP-ISUP/SS/HOLD/	
SIP selection criteria:	PICS 8/4 PICS 8/3	
ISUP selection criteria:	PICS 5/22	
Test purpose:	*A party can hold and retrieve the remote party in the confirmed state using the UPDATE method (sending)* Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. The called party should be able to put the other party on hold The called party should be able to retrieve the other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM UPDATE(sendonly) ← ← CPG(hold) 200 OK INVITE(recvonly) → UPDATE(sendrecv) ← ← CPG(retrieve) 200 OK UPDATE(recvonly) →	

<table>
<tr><th>TP502006</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
7.4.10/[14]</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/SS/HOLD/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 8/4</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 5/22</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Both parties can hold and retrieve the remote party in the confirmed state. First hold party retrieves first</i>
Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party in held state can put the remote party put on hold. Ensure that a party can retrieve the call previously put on hold.

The calling party should be able to put the other party on hold
The called party should be able to put the other party on hold
The calling party should be able to retrieve the other party
The called party should be able to retrieve the other party</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">SDP: a=sendonly or a=inactive (put on hold)
a=sendrecv or a=recvonly or omitted (retrieve the call)
o= . . <version incremented></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold)
Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>SIP</th><th></th><th>MGCF</th><th></th><th>ISUP</th></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td></td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td></td></tr>
<tr><td>INVITE(sendonly)</td><td>→</td><td></td><td>→</td><td>CPG(hold)</td></tr>
<tr><td>200 OK INVITE(recvonly)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>INVITE(inactive)</td><td>←</td><td></td><td>←</td><td>CPG(hold)</td></tr>
<tr><td>200 OK INVITE(inactive)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>INVITE(recvonly)</td><td>→</td><td></td><td>→</td><td>CPG(retrieve)</td></tr>
<tr><td>200 OK INVITE(sendonly)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>INVITE(sendrecv)</td><td>←</td><td></td><td>←</td><td>CPG(retrieve)</td></tr>
<tr><td>200 OK INVITE(sendrecv)</td><td>→</td><td></td><td></td><td></td></tr>
</table>
</td></tr>
</table>

<table>
<tr><th>TP502007</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
7.4.10/[14]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISUP/SS/HOLD/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 8/4</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5">PICS 5/22</td></tr>
<tr><td>Test purpose:</td><td colspan="5">*Both partys can hold and retrieve the remote party in the confirmed state. Second hold party retrieves first*
Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party in held state can put the remote party put on hold. Ensure that a party can retrieve the call previously put on hold.

The calling party should be able to put the other party on hold
The called party should be able to put the other party on hold
The called party should be able to retrieve the other party
The calling party should be able to retrieve the other party</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">SDP: a=sendonly or a=inactive (put on hold)
a=sendrecv or a=recvonly or omitted (retrieve the call)
o= . . <version incremented></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5">CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold)
Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)</td></tr>
<tr><td rowspan="13">Comments:</td><td>SIP</td><td></td><td>MGCF</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td></td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td></td></tr>
<tr><td>INVITE(sendonly)</td><td>→</td><td></td><td>→</td><td>CPG(hold)</td></tr>
<tr><td>200 OK INVITE(recvonly)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>INVITE(inactive)</td><td>←</td><td></td><td>←</td><td>CPG(hold)</td></tr>
<tr><td>200 OK INVITE(inactive)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>INVITE(recvonly)</td><td>←</td><td></td><td>←</td><td>CPG(retrieve)</td></tr>
<tr><td>200 OK INVITE(sendonly)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>INVITE(sendrecv)</td><td>→</td><td></td><td>→</td><td>CPG(retrieve)</td></tr>
<tr><td>200 OK INVITE(sendrecv)</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td></tr>
</table>

6.3.1.3 Terminal portability (TP)

Void.

6.3.1.4 Conference calling (CONF)

<table>
<tr><th>TP504001</th><th>SIP reference: RFC 3261 [6]</th><th>NGN reference:
ES 283 027 [1], clause 7.4.14</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/SS/CONF/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 8/2</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 5/10</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Generic notification Conference established and Conference disconnected and SIP procedure</i>
Ensure that the SUT does not stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator was received due to the CONF supplementary service.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">CPG: Generic notification = Conference established
CPG: Generic notification = Conference disconnected</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>SIP</th><th></th><th>SUT</th><th></th><th>ISUP</th></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>Ringing tone</td><td></td><td></td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>CPG(Conference established)</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>CPG(Conference disconnected)</td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>
</td></tr>
</table>

TP504002	**SIP reference: RFC 3261 [6]**	**NGN reference:** **ES 283 027 [1], clause 7.4.14**
TSS reference:	SIP-ISUP/SS/CONF/	
SIP selection criteria:	PICS 8/2	
ISUP selection criteria:	PICS 5/10	
Test purpose:	*Generic notification Isolated and Reattached and SIP procedure* Ensure that the SUT stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value **GEN_NOT_VALUE** was received due to the CONF supplementary service. If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute line a=**sendonly/sendrecv** or omitted attribute line, else: no mapping.	
SIP Parameter values:	SDP: a=**sendonly/sentrecv** or a line is omitted	
ISUP Parameter values:	CPG: Generic notification = Conference established CPG: Generic notification = **GEN_NOT_VALUE** CPG: Generic notification = Conference disconnected	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation ← CPG(Conference established) INVITE(sendonly) ← ← CPG(Isolated) 200 OK INVITE(recvonly) → ACK ← INVITE(sendrecv) ← ← CPG(Reattached) 200 OK INVITE(sendrecv) → ACK ← Conversation ← CPG(Conference disconnected) BYE → → REL 200 OK BYE ← ← RLC	

TP504003	**SIP reference: RFC 3261 [6]**	**NGN reference: ES 283 027 [1], clause 7.4.14**
TSS reference:	SIP-ISUP/SS/CONF/	
SIP selection criteria:	NOT PICS 5/10	
ISUP selection criteria:		
Test purpose:	*No mapping of isolated and reattached* Ensure that the SUT on receipt of a CPG message due to the CONF supplementary service, the Generic notification indicator with the value. **No mapping, no disrupting the SIP procedure.**	
SIP Parameter values:	No mapping	
ISUP Parameter values:	CPG: Generic notification = Conference established CPG: Generic notification = isolated CPG: Generic notification = reattached CPG: Generic notification = Conference disconnected	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation ← CPG(Conference established) ← CPG(Isolated) ← CPG(Reattached) Conversation ← CPG(Conference disconnected) BYE → → REL 200 OK BYE ← ← RLC	

TP504004	**SIP reference: RFC 3261 [6]**	**NGN reference: 7.4.14/[14]**
TSS reference:	SIP-ISUP/SS/CONF/	
SIP selection criteria:	PICS [16] 8/2	
ISUP selection criteria:	PICS [16] 5/10	
Test purpose:	*No mapping of generic notifications no change the session state* Ensure that the MGCF can receive in a CPG the Generic notifications is "other party added" or "other party isolated" or "other party reattached" or "other party split" or " conference floating" or " other party disconnected" and there is no mapping on the SIP side and the call is not disrupted.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → ← CPG(Conference established) ← CPG(other party added) ← CPG(other party isolated) ← CPG(other party reattached) ← CPG(other party split) ← CPG(other party disconnected) ← CPG(Conference floating) ← CPG(Conference disconnected) BYE ← ← REL 200 OK BYE → → RLC	

TP504005	**SIP reference: RFC 3261 [6]**	**NGN reference: ES 283 027 [1], clause 7.5.6**
TSS reference:	SIP-ISUP/SS/CONF/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "conference established"* Upon the receipt of a conference information document with the <conference-state-type> element active is set to "true", the MGCF shall send a CPG message to the CS side with a notification "**conference established**".	
SIP Parameter values:	NOTIFY 1: Event contains **conference**; Subscription-State contains **active; expires=xxxx** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>2</user-count> if present <active>**true**</active> if present <*users*> <*user* entity=ISUPx URI state="full" <endpoint entity=endpoint ISUPx URI <status>connected</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:		
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → INVITE(SDP focus) → 200 OK INVITE ← ACK → NOTIFY 1 → → CPG(Conference established) BYE ← ← REL 200 OK BYE → → RLC	

<table>
<tr><th>TP504006</th><th>SIP reference: RFC 3261 [6]</th><th>NGN reference:
ES 283 027 [1], clause 7.5.6</th></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/SS/CONF/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 1/1</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Conference notification information is mapped into "other party added"</i>

Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to "connected" and it was not set to "on-hold" before and the Contact URI in the element entity is not the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "other other party added".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">NOTIFY 1: see test case 504005
NOTIFY 2: Event contains conference; Subscription-State contains active
application/conference-info+xml:
<<i>conference-info</i>>
entity=conference URI state="full" version="x"
<<i>conference-state</i>>
<user-count>3</user-count> if present
<<i>users</i>>
<<i>user</i> entity=SIPx URI state="full"
<endpoint entity=endpoint SIPx URI
<status>connected</status>
<joining-method>dialed-out</ joining-method>
<media id="1"
<status>sendrecv</status></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2"></td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>SIP</th><th></th><th>MGCF</th><th></th><th>ISUP</th></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>INVITE(SDP focus)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>NOTIFY 1</td><td>→</td><td></td><td>→</td><td>CPG(Conference established)</td></tr>
<tr><td>200 OK NOTIFY</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>NOTIFY 2</td><td>→</td><td></td><td>→</td><td>CPG(other party added)</td></tr>
<tr><td>200 OK NOTIFY</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>RLC</td></tr>
</table>
</td></tr>
</table>

TP504007	**SIP reference: RFC 3261 [6]**	**NGN reference:** **ES 283 027 [1], clause 7.5.6**
TSS reference:	SIP-ISUP/SS/CONF/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "isolated"* Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to "on-hold" and it was set to "connected" before and the Contact URI in the element entity is the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "**isolated**".	
SIP Parameter values:	NOTIFY 1: see test case 504005 NOTIFY 2: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>3</user-count> if present <*users*> <*user* entity=ISUPx URI state="full" <endpoint entity=endpoint ISUPx URI <status>**on-hold**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:		
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → INVITE(SDP focus) → 200 OK INVITE ← ACK → NOTIFY 1 → → CPG(Conference established) 200 OK NOTIFY ← NOTIFY 2 → → CPG(isolated) 200 OK NOTIFY ← BYE ← ← REL 200 OK BYE → → RLC	

<table>
<tr><td>TP504008</td><td colspan="3">SIP reference: RFC 3261 [6]</td><td colspan="2">NGN reference:
ES 283 027 [1], clause 7.5.6</td></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISUP/SS/CONF/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 1/1</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>Conference notification information is mapped into "other party isolated"</i>

Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to "on-hold" and it was set to "connected" before and the Contact URI in the element entity is not the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "other party isolated".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">NOTIFY 1: see test case 504005
NOTIFY 2: Event contains conference; Subscription-State contains active
application/conference-info+xml:
<<i>conference-info</i>>
entity=conference URI state="full" version="x"
<<i>conference-state</i>>
<user-count>3</user-count> if present
<<i>users</i>>
<<i>user</i> entity=SIPx URI state="full"
<endpoint entity=endpoint SIPx URI
<status>on-hold</status>
<joining-method>dialed-out</ joining-method>
<media id="1"
<status>sendrecv</status></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5"></td></tr>
<tr><td>Comments:</td><td>SIP</td><td></td><td>MGCF</td><td></td><td>ISUP</td></tr>
<tr><td></td><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td></td><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td></td><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td></td><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td>INVITE(SDP focus)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td>200 OK INVITE</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td>NOTIFY 1</td><td>→</td><td></td><td>→</td><td>CPG(Conference established)</td></tr>
<tr><td></td><td>200 OK NOTIFY</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td>NOTIFY 2</td><td>→</td><td></td><td>→</td><td>CPG(other party isolated)</td></tr>
<tr><td></td><td>200 OK NOTIFY</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td>BYE</td><td>←</td><td></td><td>←</td><td>REL</td></tr>
<tr><td></td><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>RLC</td></tr>
</table>

TP504009	**SIP reference: RFC 3261 [6]**	**NGN reference: ES 283 027 [1], clause 7.5.6**
TSS reference:	SIP-ISUP/SS/CONF/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "reattached"* Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to "connected" and it was set to "on-hold" before and the Contact URI in the element entity is the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "**reattached**".	
SIP Parameter values:	NOTIFY 1: see test case 504005 NOTIFY 2: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: *<conference-info>* entity=conference URI state="full" version="x" *<conference-state>* <user-count>3</user-count> if present *<users>* *<user* entity=ISUPx URI state="full" <endpoint entity=endpoint ISUPx URI <status>**on-hold**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status> NOTIFY 3: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: *<conference-info>* entity=conference URI state="full" version="x" *<conference-state>* <user-count>3</user-count> if present *<users>* *<user* entity=ISUPx URI state="full" <endpoint entity=endpoint ISUPx URI <status>**connected**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:		
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → INVITE(SDP focus) → 200 OK INVITE ← ACK → NOTIFY 1 → → CPG(Conference established) 200 OK NOTIFY ← NOTIFY 2 → → CPG(isolated) 200 OK NOTIFY ← NOTIFY 3 → → CPG(reattached) 200 OK NOTIFY ← BYE ← ← REL 200 OK BYE → → RLC	

TP504010	**SIP reference: RFC 3261 [6]**	**NGN reference:** **ES 283 027 [1], clause 7.5.6**
TSS reference:	SIP-ISUP/SS/CONF/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "other party reattached"* Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to "connected" and it was set to "on-hold" before and the Contact URI in the element entity is not the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "**other party reattached**".	
SIP Parameter values:	NOTIFY 1: see test case 504005 NOTIFY 2: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>3</user-count> if present <*users*> <*user* entity=SIPx URI state="full" <endpoint entity=endpoint SIPx URI <status>**on-hold**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status> NOTIFY 3: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>3</user-count> if present <*users*> <*user* entity=SIPx URI state="full" <endpoint entity=endpoint SIPx URI <status>**connected**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:		
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → INVITE(SDP focus) → 200 OK INVITE ← ACK → NOTIFY 1 → → CPG(Conference established) 200 OK NOTIFY ← NOTIFY 2 → → CPG(other party isolated) 200 OK NOTIFY ← NOTIFY 3 → → CPG(other party reattached) 200 OK NOTIFY ← BYE ← ← REL 200 OK BYE → → RLC	

TP504011	**SIP reference: RFC 3261 [6]**	**NGN reference:** **ES 283 027 [1], clause 7.5.6**
TSS reference:	SIP-ISUP/SS/CONF/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "other party disconnected"* Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to "disconnected" and the element joining-method of joining-type is not set to "focus-owner, the MGCF shall send a CPG message to the CS side with a notification "**other party disconnected**".	
SIP Parameter values:	NOTIFY 1: see test case 504005 NOTIFY 2: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: *<conference-info>* entity=conference URI state="full" version="x" *<conference-state>* <user-count>3</user-count> if present *<users>* *<user* entity=SIPx URI state="full" <endpoint entity=endpoint SIPx URI <status>**connected**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status> NOTIFY 3: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: *<conference-info>* entity=conference URI state="full" version="x" *<conference-state>* <user-count>3</user-count> if present *<users>* *<user* entity=SIPx URI state="full" <endpoint entity=endpoint SIPx URI <status>**disconnected**</status> <joining-method>dialed-out</ joining-method> <disconnection-method>departed<disconnection-method/> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:		
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → INVITE(SDP focus) → 200 OK INVITE ← ACK → NOTIFY 1 → → CPG(Conference established) 200 OK NOTIFY ← NOTIFY 2 → → CPG(other party added) 200 OK NOTIFY ← NOTIFY 3 → → CPG(other party disconnected) 200 OK NOTIFY ← BYE ← ← REL 200 OK BYE → → RLC	

TP504012	**SIP reference: RFC 3261 [6]**	**NGN reference:** **ES 283 027 [1], clause 7.5.6**
TSS reference:	SIP-ISUP/SS/CONF/	
SIP selection criteria:	NOT PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "other party added"* Upon the receipt of a conference information document the conference notification information is not mapped to the PSTN side. No NOTIFY is sent to the ISDN user.	
SIP Parameter values:	NOTIFY 1: see test case 504005 NOTIFY 2: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: *<conference-info>* entity=conference URI state="full" version="x" *<conference-state>* <user-count>3</user-count> if present *<users>* *<user* entity=SIPx URI state="full" <endpoint entity=endpoint SIPx URI <status>**connected**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:		
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM INVITE(SDP focus) → 200 OK INVITE ← ACK → NOTIFY 1 → → CPG(Conference established) 200 OK NOTIFY ← NOTIFY 2 → → CPG(other party added) 200 OK NOTIFY ← BYE ← ← REL 200 OK BYE → → RLC	

TP504013	**SIP reference: RFC 3261 [6]**	**NGN reference:** **ES 283 027 [1], clause 7.5.6**
TSS reference:	SIP-ISUP/SS/CONF/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*The referring of MGCF is not possible call is established* Ensure that a REFER request received by the MGCF is not successful. The request is rejected with 403 Forbidden. The CS -site is not affected.	
SIP Parameter values:	REFER: Request URI contained the **conference URI** Refer-To contains the URI of **ISUPx**, method=invite Referred-By contains SIP or tel URI of **SIPx**	
ISUP Parameter values:		
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → REFER → 403 Forbidden ←	

TP504014	**SIP reference: RFC 3261 [6]**	**NGN reference:** **ES 283 027 [1], clause 7.5.6**
TSS reference:	SIP-ISUP/SS/CONF/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*The referring of MGCF is not possible call is not established* Ensure that a REFER request received by the MGCF is not successful. The request is rejected with 403 Forbidden. The CS -site is not affected.	
SIP Parameter values:	REFER: Request URI contained the **conference URI** Refer-To contains the URI of **ISUPx**, method=invite Referred-By contains SIP or tel URI of **SIPx**	
ISUP Parameter values:		
Comments:	**SIP** **MGCF** **ISUP** REFER → 403 Forbidden ←	

6.3.1.5 Three Party service (3PTY)

TP505001	**SIP reference: RFC 3261 [6]**	**NGN reference: ES 283 027 [1], clause 7.4.15**
TSS reference:	SIP-ISUP/SS/3PTY/	
SIP selection criteria:	PICS 8/2	
ISUP selection criteria:	PICS 5/5 AND PICS 5/18	
Test purpose:	*Notification procedure supported* Ensure that the SUT stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value **GEN_NOT_VALUE** was received due to the 3PTY supplementary service. • If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute line **a_LINE_VA**, or omitted attribute line, else: no mapping.	
SIP Parameter values:	SDP: a= **a_LINE_VA** (see table 38)	
ISUP Parameter values:	CPG: notification = remote hold CPG: Generic notification = **GEN_NOT_VALUE**	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation INVITE(sendonly) ← ← CPG(hold) 200 OK INVITE(recvonly) → ACK ← INVITE(sendrecv) ← ← CPG(Conference established) 200 OK INVITE(sendrecv) → ACK ← Conversation ← CPG(Conference disconnected) Conversation BYE → → REL 200 OK BYE ← ← RLC	

Table 38

Values for test purpose TP505001		
←INVITE/UPDATE SDP attribute line a_LINE_VA		**← CPG Generic notification GEN_NOT_VALUE**
VA_01	sendonly or inactive	Conference established
VA_02	sendrecv or recvonly or omitted	Conference disconnected

TP505002	**SIP reference: RFC 3261 [6]**	**NGN reference: ES 283 027 [1], clause 7.4.15 ITU-T Rec Q.734.2 [35], clause 2.7**
TSS reference:	SIP-ISUP/SS/3PTY/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 5/18	
Test purpose:	*Notification procedure not supported* Ensure that the SUT on receipt of a CPG message due to the 3PTY supplementary service, the Generic notification indicator with the value. No mapping, no disrupting the SIP procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	CPG: Generic notification = Conference established CPG: Generic notification = Conference disconnected	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone ← CPG(hold) ← CPG(Conference established) Conversation ← CPG(Conference disconnected) Conversation BYE → → REL 200 OK BYE ← ← RLC	

TP504003	**SIP reference: RFC 3261 [6]**	**NGN reference: ES 283 027 [1], clause 7.4.15**
TSS reference:	SIP-ISUP/SS/3PTY	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "conference established"* Upon the receipt of a conference information document with the <conference-state-type> element active is set to "true", the MGCF shall send a CPG message to the CS side with a notification "**conference established**".	
SIP Parameter values:	NOTIFY 1: Event contains **conference**; Subscription-State contains **active; expires=xxxx** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>2</user-count> if present <active>**true**</active> if present <*users*> <*user* entity=ISUPx URI state="full" <endpoint entity=endpoint ISUPx URI <status>connected</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:		
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → INVITE(sendonly) → → CPG(hold) 200 OK INVITE(recvonly) ← ACK → INVITE(SDP focus) → 200 OK INVITE ← ACK → NOTIFY 1 → → CPG(Conference established) BYE ← ← REL 200 OK BYE → → RLC	

6.3.1.6 Connected line identification (COL)

TP506001		**ISUP reference:** **[15] clause 7.4.2**
TSS reference:	SIP-ISUP/SS/COL/	
SIP selection criteria:	NOT PICS 5/22	
ISUP selection criteria:		
Test purpose:	*Mapping of connected number not supported* Ensure that the SUT, if a connected number is received in an ANM, does not disrupt the SIP signalling procedure. The connected number is not mapped into any SIP message.	
SIP Parameter values:		
ISUP Parameter values:	ANM: Connected number Parameter	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Conversation** BYE → → REL 200 OK BYE ← ← RLC	

TP506002		**ISUP reference:** **[14] clauses 7.4.2 and 7.5.2**
TSS reference:	SIP-ISUP/SS/COL/	
SIP selection criteria:	PICS 5/22 AND PICS 13/1	
ISUP selection criteria:		
Test purpose:	*Connected number national, presentation allowed, no additional connected number received* Ensure that the SUT, on receipt of an ANM message with a **Connected number parameter** coded Address presentation restricted parameter = presentation allowed Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals in the format: NDC+SN and without the **Generic number parameter**, sends a 200 OK INVITE to the UAC with a P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and Add CC (of the country where the MGCF is located) to Connected PN address signals to construct E.164 number in URI global number format.	
SIP Parameter values:	200 OK INVITE: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN	
ISUP Parameter values:	ANM; **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = Network provided Address signals = derrived from the P-Asserted-Identity **Generic number parameter** not present	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Conversation** BYE → → REL 200 OK BYE ← ← RLC	

TP506003		**NGN reference:** **ES 283 027 [1], clauses 7.4.2 and 7.5.2**
TSS reference:	SIP-ISUP/SS/COL/	
SIP selection criteria:	PICS 5/22 AND PICS 13/1	
ISUP selection criteria:		
Test purpose:	*Connected number international, presentation allowed, no additional connected number received* Ensure that the SUT, on receipt of an ANM message with a **Connected number parameter** coded Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals in the format: CC+NDC+SN and without the **Generic number parameter**, sends a 200 OK INVITE to the UAC with a P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN as received in the connected number in the ANM.	
SIP Parameter values:	200 OK INVITE: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN	
ISUP Parameter values:	ANM; **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = 0000100'B Numbering plan indicator = '001'B Screening indicator = Network provided Address signals = PIXIT **Generic number parameter** not present	
Comments:	**SIP** / **MGCF** / **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Conversation** BYE → → REL 200 OK BYE ← ← RLC	

<table>
<tr><td>TP506004</td><td></td><td>ISUP reference:
[14] clauses 7.4.2.2 and 7.5.2</td></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/SS/COL/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 5/22 AND NOT PICS 13/1</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Connected number national, presentation allowed, additional connected number received</i>

Ensure that the SUT, on receipt of an ANM message with a

Connected number parameter coded
Address presentation restricted parameter = presentation allowed
Nature of address indicator = national number
Numbering plan indicator = ISDN/Telephony numbering plan
Screening indicator = Network provided
Address signals in the format: NDC+SN

Generic number parameter,
Number Qualifier Indicator <i>"Additional connected number"</i>
Address presentation restricted parameter = presentation allowed
Nature of address indicator = national number
Numbering plan indicator = ISDN/Telephony numbering plan
Screening indicator = user provided, not verified
Address signals = PIXIT NDC+SN

sends a 200 OK INVITE to the UAC with a

P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and Add CC (of the country where the MGCF is located) to Connected PN address signals to construct E.164 number in URI. Prefix number with "+".
The additional connected number is not interworked</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">200 OK INVITE: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">ANM;
Connected number parameter
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = Network provided
Address signals = PIXIT
Generic number parameter
Number Qualifier Indicator "00000101"B
Address presentation restricted parameter = '00'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = '00'B
Address signals = PIXIT</td></tr>
<tr><td>Comments:</td><td colspan="2">SIP MGCF ISUP
INVITE → → IAM
180 Ringing ← ← ACM
200 OK INVITE ← ← ANM
ACK →
Conversation
BYE → → REL
200 OK BYE ← ← RLC</td></tr>
</table>

TP506005		**ISUP reference:** **[14] clauses 7.4.2.2 and 7.5.2**
TSS reference:	SIP-ISUP/SS/COL/	
SIP selection criteria:	PICS 5/22 AND NOT PICS 13/1	
ISUP selection criteria:		
Test purpose:	*Connected number international, presentation allowed, additional connected number received* Ensure that the SUT, on receipt of an ANM message with a **Connected number parameter** coded Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals in the format: PIXIT CC+NDC+SN **Generic number parameter**, Number Qualifier Indicator *"Additional connected number"* Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = CC+NDC+SN sends a 200 OK INVITE to the UAC with a P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and the The additional connected number is not interworked	
SIP Parameter values:	200 OK INVITE: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN	
ISUP Parameter values:	ANM; **Connected number parameter** Address presentation restricted parameter = '00'B Nature of address indicator = "0000100'B Numbering plan indicator = '001'B Screening indicator = Network provided Address signals = PIXIT **Generic number parameter** Number Qualifier Indicator "00000101"B Address presentation restricted parameter = '00'B Nature of address indicator = '0000100'B Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT	
Comments:		

SIP		**MGCF**		**ISUP**
INVITE	→		→	IAM
180 Ringing	←		←	ACM
200 OK INVITE	←		←	ANM
ACK	→			
		Conversation		
BYE	→		→	REL
200 OK BYE	←		←	RLC

<table>
<tr><td>TP506006</td><td></td><td>ISUP reference:
[14] clauses 7.4.2 and 7.5.2</td></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/SS/COL/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 5/22 AND NOT PICS 13/1</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Connected number national, presentation restricted, no additional connected number received</i>

Ensure that the SUT, on receipt of an ANM message with a

Connected number parameter coded
Address presentation restricted parameter = presentation restricted
Nature of address indicator = national number
Numbering plan indicator = ISDN/Telephony numbering plan
Screening indicator = Network provided
Address signals in the format: PIXIT NDC+SN

and without the Generic number parameter,

sends a 200 OK INVITE to the UAC with a

P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and Add CC (of the country where the MGCF is located) to Connected PN address signals to construct E.164 number in URI. Prefix number with "+".
a Privacy header is inserted with the value "id" or the value "id" is added to a existence Privacy header</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">200 OK INVITE: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">ANM;
Connected number parameter
Address presentation restricted parameter = '01'B
Nature of address indicator = '0000011'B
Numbering plan indicator = '001'B
Screening indicator = Network provided
Address signals = PIXIT
Generic number parameter not present</td></tr>
<tr><td>Comments:</td><td colspan="2"><table>
<tr><td>SIP</td><td></td><td>MGCF</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table></td></tr>
</table>

TP506007		**ISUP reference:** **[14] clauses 7.4.2 and 7.5.2**
TSS reference:	SIP-ISUP/SS/COL/	
SIP selection criteria:	PICS 5/22 AND NOT PICS 13/1	
ISUP selection criteria:		
Test purpose:	*Connected number international, presentation restricted, no additional connected number received* Ensure that the SUT, on receipt of an ANM message with a **Connected number parameter** coded Address presentation restricted parameter = presentation restricted Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals in the format: PIXIT CC+NDC+SN and without the **Generic number parameter**, sends a 200 OK INVITE to the UAC with a P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and the a Privacy header is inserted with the value "id" or the value "id" is added to a existence Privacy header	
SIP Parameter values:	200 OK INVITE: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN	
ISUP Parameter values:	ANM; **Connected number parameter** Address presentation restricted parameter = '01'B Nature of address indicator = 0000100'B Numbering plan indicator = '001'B Screening indicator = Network provided Address signals = PIXIT **Generic number parameter** not present	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Conversation** BYE → → REL 200 OK BYE ← ← RLC	

TP506008		**ISUP reference:** **[14] clauses 7.4.2 and 7.5.2**
TSS reference:	SIP-ISUP/SS/COL/	
SIP selection criteria:	PICS 5/22 AND NOT PICS 13/1	
ISUP selection criteria:		
Test purpose:	*Connected number national, presentation restricted, additional connected number received* Ensure that the SUT, on receipt of an ANM message with a **Connected number parameter** coded Address presentation restricted parameter = presentation restricted Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals in the format: PIXIT NDC+SN **Generic number parameter**, Number Qualifier Indicator *"Additional connected number"* Address presentation restricted parameter = presentation restricted Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = NDC+SN sends a 200 OK INVITE to the UAC with a P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and Add CC (of the country where the MGCF is located) to Connected PN address signals to construct E.164 number in URI. Prefix number with "+". a Privacy header is inserted with the value "id" or the value "id" is added to a existence Privacy header The additional connected number is not interworked	
SIP Parameter values:	200 OK INVITE: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN	
ISUP Parameter values:	ANM; **Connected number parameter** Address presentation restricted parameter = '01'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = Network provided Address signals = PIXIT **Generic number parameter** Number Qualifier Indicator "00000101"B Address presentation restricted parameter = '01'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT	

Comments:	**SIP**		**MGCF**		**ISUP**
	INVITE	→		→	IAM
	180 Ringing	←		←	ACM
	200 OK INVITE	←		←	ANM
	ACK	→			
			Conversation		
	BYE	→		→	REL
	200 OK BYE	←		←	RLC

TP506009		**ISUP reference:** **[14] clauses 7.4.2 and 7.5.2**
TSS reference:	SIP-ISUP/SS/COL/	
SIP selection criteria:	PICS 5/22 AND NOT PICS 13/1	
ISUP selection criteria:		
Test purpose:	*Connected number international, presentation restricted, additional connected number received* Ensure that the SUT, on receipt of an ANM message with a **Connected number parameter** coded Address presentation restricted parameter = presentation restricted Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals in the format: PIXIT CC+NDC+SN **Generic number parameter**, Number Qualifier Indicator *"Additional connected number"* Address presentation restricted parameter = presentation restricted Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = CC+NDC+SN sends a 200 OK INVITE to the UAC with a P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and the a Privacy header is inserted with the value "id" or the value "id" is added to a existence Privacy header The additional connected number is not interworked	
SIP Parameter values:	200 OK INVITE: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN	
ISUP Parameter values:	ANM; **Connected number parameter** Address presentation restricted parameter = '01'B Nature of address indicator = "0000100'B Numbering plan indicator = '001'B Screening indicator = Network provided Address signals = PIXIT **Generic number parameter** Number Qualifier Indicator "00000101"B Address presentation restricted parameter = '001B Nature of address indicator = '0000100'B Numbering plan indicator = '001'B Screening indicator = '00'B Address signals = PIXIT	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Conversation** BYE → → REL 200 OK BYE ← ← RLC	

Values for test purposes TP102006-TP102009	
VA_01	ISUP_SI = user provided verified and passed, '01'B
VA_02	ISUP_SI = network provided, '11'B

<table>
<tr><th>TP506010</th><th colspan="2"></th><th colspan="3">ISUP reference:
[14] clauses 7.4.2 and 7.5.2</th></tr>
<tr><td>TSS reference:</td><td colspan="5">SIP-ISUP/SS/COL/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 5/22 AND PICS 13/1</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>IAM connected line request indication is sent</i>
Ensure that a optional forward call indicator value Connected line identity request indicator is set to "requested" is contained in the sent IAM if an INVITE request is received containing a Supported header equal to "from-change".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">INVITE: Supported: "from-change"</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5">IAM: oFCi Connected line identity request indicator is set to "requested"</td></tr>
<tr><td rowspan="8">Comments:</td><td>SIP</td><td></td><td>MGCF</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td colspan="5">Conversation</td></tr>
<tr><td>BYE</td><td>→</td><td></td><td>→</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>←</td><td></td><td>←</td><td>RLC</td></tr>
</table>

TP506011		**ISUP reference:** **[14] clauses 7.4.2 and 7.5.2**
TSS reference:	SIP-ISUP/SS/COL/	
SIP selection criteria:	PICS 5/22 AND PICS 13/1	
ISUP selection criteria:		
Test purpose:	*Additional connected number national, presentation allowed in ANM is received* Ensure that if a ANM is received and a Additional connected number "national number", "prestentation allowed" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag.	
SIP Parameter values:	200 OK INVITE: Supported: "from-change" P-Asserted-Identity derived from the Connected number UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN	
ISUP Parameter values:	IAM: oFCi Connected line identity request indicator is set to "requested" ANM: **Additional connected number** Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT **Connected number parameter** Address presentation restricted parameter = presentation allowed Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → UPDATE ← 200 OK UPDATE → **Conversation** BYE → → REL 200 OK BYE ← ← RLC	

TP506012		**ISUP reference:** **[14] clauses 7.4.2 and 7.5.2**
TSS reference:	SIP-ISUP/SS/COL/	
SIP selection criteria:	PICS 5/22 AND PICS 13/1	
ISUP selection criteria:		
Test purpose:	*Additional connected number international, presentation allowed in ANM is received* Ensure that if a ANM is received and a Additional connected number "international number" "prestentation allowed" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag.	
SIP Parameter values:	200 OK INVITE: Supported: "from-change" P-Asserted-Identity derived from the Connected number UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN	
ISUP Parameter values:	IAM: oFCi Connected line identity request indicator is set to "requested" ANM: **Additional connected number** Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT **Connected number parameter** Address presentation restricted parameter = presentation allowed Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → UPDATE ← 200 OK UPDATE → **Conversation** BYE → → REL 200 OK BYE ← ← RLC	

TP506013		**ISUP reference:** **[14] clauses 7.4.2 and 7.5.2**
TSS reference:	SIP-ISUP/SS/COL/	
SIP selection criteria:	PICS 5/22 AND PICS 13/1	
ISUP selection criteria:		
Test purpose:	*Additional connected number national, presentation restricted in ANM is received* Ensure that if a ANM is received and a Additional connected number "national number" "prestentation restricted" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag.	
SIP Parameter values:	200 OK INVITE: Supported: "from-change" P-Asserted-Identity derived from the Connected number in the format "+"CC+NDC+SN Privacy: id UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN Privacy: header	
ISUP Parameter values:	IAM: oFCi Connected line identity request indicator is set to "requested" ANM: **Additional connected number** Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation restricted Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT **Connected number parameter** Address presentation restricted parameter = presentation restricted Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → UPDATE ← 200 OK UPDATE → **Conversation** BYE → → REL 200 OK BYE ← ← RLC	

TP506014		**ISUP reference:** **[14] clauses 7.4.2 and 7.5.2**
TSS reference:	SIP-ISUP/SS/COL/	
SIP selection criteria:	PICS 5/22 AND PICS 13/1	
ISUP selection criteria:		
Test purpose:	*Additional connected number international, presentation restricted in ANM is received* Ensure that if a ANM is received and a Additional connected number "international number" "prestentation restricted"is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag.	
SIP Parameter values:	200 OK INVITE: Supported: "from-change" P-Asserted-Identity derived from the Connected number in the format "+"CC+NDC+SN Privacy: id UPDATE: From header contains the Generic number in the format "+"CC+NDC+SN Privacy: header	
ISUP Parameter values:	IAM: oFCi Connected line identity request indicator is set to "requested" ANM: **Additional connected number** Number Qualifier Indicator "00000101"B Address presentation restricted parameter = presentation restricted Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = user provided, not verified Address signals = PIXIT **Connected number parameter** Address presentation restricted parameter = presentation restricted Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network provided Address signals = PIXIT	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → UPDATE ← 200 OK UPDATE → **Conversation** BYE → → REL 200 OK BYE ← ← RLC	

6.3.1.7 Malicious call identification MCID

TP507001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.4**
TSS reference:	SIP-ISUP/SS/MCID/	
SIP selection criteria:	PICS 9/1	
ISUP selection criteria:		
Test purpose:	*No interworking MGCF sends IRS* Ensure that the SUT if an IDR is received returns an IRS message. The MCID response indicator is set to "MCID not included". The SIP signalling procedure is not disrupted.	
SIP Parameter values:	No influence	
ISUP Parameter values:	IDR: MCID requested IRS: MCID not included	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM ← IDR → IRS 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP507002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.4**
TSS reference:	SIP-ISUP/SS/MCID/	
SIP selection criteria:	NOT PICS 9/1	
ISUP selection criteria:		
Test purpose:	*No interworking timeout T39* Ensure that the SUT if an IDR is received, no IDR is sent. The SIP signalling procedure is not disrupted.	
SIP Parameter values:	No influence	
ISUP Parameter values:	IDR: MCID requested	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM ← IDR T39 timeout 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

6.3.1.8 Sub-addressing (SUB)

TP508001	**SUB Reference:** **ES 283 027 [1], clause 7.4.5**	**Selection criteria:** **PICS 5/8**
TSS reference:	SIP-ISUP/SS/SUB/	
Preconditions:		
Test purpose:	*The isub parameter of the P-Asserted-Identity header in an INVITE is mapped in the calling party subaddress in the IAM* Ensure that the isub parameter in the P-Asserted-Identity header of the received INVITE is interworked in the Calling party subaddress contained in an ATP parameter in the sent IAM. The Type of Subbaddress is set set to "0 0 0" "NSAP (ITU-T Recommendation X.213 [29] and ISO/IEC 8348 [30] Add.2)"	
SIP Parameter values:	INVITE: P-Asserted-Identity: sip: user part; isub=<subaddress>@hostportion	
ISUP Parameter values:	IAM: ATP(Calling party subaddress)	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 100 Trying ← 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → Communication BYE → → REL 200 OK BYE ← ← RLC	

TP508002	**SUB Reference:** **ES 283 027 [1], clause 7.4.5**	**Selection criteria:** PICS 5/8
TSS reference:	SIP-ISUP/SS/SUB/	
Preconditions:		
Test purpose:	*The isub parameter of the Request URI in an INVITE is mapped in the called party subaddress in the IAM* Ensure that the isub parameter in the Request URI of the received INVITE is interworked is the Called party subaddress contained in an ATP parameter in the sent IAM. The Type of Subbaddress is set set t "0 0 0" "NSAP (ITU-T Recommendation X.213 [29] and ISO/IEC 8348 [30] Add.2)"	
SIP Parameter values:	INVITE: sip: user part; isub=<subaddress>@hostportion	
ISUP Parameter values:	IAM: ATP(Called party subaddress)	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 100 Trying ← 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → Communication BYE → → REL 200 OK BYE ← ← RLC	

TP508003	**SUB Reference:** **ES 283 027 [1], clause 7.4.5**	**Selection criteria:** **PICS 5/8**
TSS reference:	SIP-ISUP/SS/SUB/	
Preconditions:		
Test purpose:	*The connected subaddress in the ANM is mapped in the isub parameter of the P-Asserted-Identity header in the 200 OK INVITE* Ensure that the isub parameter in the P-Asserted-Identity header of the received 200 OK INVITE is interworked in the connected subaddress contained in an ATP parameter in the sent ANM. The Type of Subbaddress is set set to "0 0 0" "NSAP (ITU-T Recommendation X.213 [29] and ISO/IEC 8348 [30] Add.2)"	
SIP Parameter values:	200 OK INVITE: P-Asserted-Identity: sip: user part; isub=<subaddress>@hostportion	
ISUP Parameter values:	IAM: oFCi: connected line request ANM: ATP(Connected subaddress)	
Comments:	**SIP** **MGCF** **ISUP** INVITE → → IAM 100 Trying ← 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → Communication BYE → → REL 200 OK BYE ← ← RLC	

6.3.1.9 Call diversion (CDIV)

TP509001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.6**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*OBCI "call diversion may occur" in ACM received no mapping* Ensure that the SUT if an ACM is received with and **call diversion may occur** indicator in the optional backward call indicator is set to "call diversion may occur", the SIP signalling procedure is not disrupted (CDa, CFNR).	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM optional backward call indicator **call diversion may occur**	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM ← ACM 180 Ringing ← ← CPG Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP509002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.6**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	NOT PICS 5/12 AND NOT PICS 5/13 AND NOT PICS 5/14 AND NOT PICS 5/15	
ISUP selection criteria:		
Test purpose:	*BCI called party status "no indication" in ACM received no mapping* Ensure that the SUT if a ACM is received called party status indicator "no indication" and containing a **Redirection number, call diversion information, redirection number restriction and generic notification set to "Call is diverting"**, the SIP signalling procedure is not disrupted (CFU, CFB, Cdi).	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: Redirection number, Call diversion information, Redirection number restriction, Generic notification	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM ← ACM 180 Ringing ← ← CPG Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP509003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.6**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	NOT PICS 5/12 AND NOT PICS 5/13 AND NOT PICS 5/14 AND NOT PICS 5/15	
ISUP selection criteria:		
Test purpose:	*CPG PROGRESS with Redirection number, Call diversion information and Generic notification received, no mapping* Ensure that the SUT if a CPG is received containing a **Redirection number, call diversion information, redirection number restriction and generic notification set to "Call is diverting"**, the SIP signalling procedure is not disrupted (CDa, CFNR, subsequent redirection).	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: Called party status "Subscriber free" CPG: Redirection number, Call diversion information, Generic notification	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM ← ACM 180 Ringing ← ← CPG Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP509004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.6**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	NOT PICS 5/12 AND NOT PICS 5/13 AND NOT PICS 5/14 AND NOT PICS 5/15	
ISUP selection criteria:		
Test purpose:	*Redirection number restriction received in ANM no mapping* Ensure that the SUT if an ANM is received with **redirection number restriction parameter**, the SIP signalling procedure is not disrupted.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ANM: Redirection number restriction	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM ← ACM 180 Ringing ← ← CPG Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP509005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	PICS 10/7	
ISUP selection criteria:		
Test purpose:	*BCI called party status "no indication" in ACM received, mapping of Redirection reason.* Ensure that the SUT, on receipt of an **ACM** message indicating a first diversion with the **Backward call indicators parameter** coded Called party's status indicator = no indication the **Call diversion information parameter** coded Notification subscription option = "010"B Redirection reason = **ISUP_REASON** and the **Generic notification indicator parameter** coded Notification indicator = call is diverting, **Redirection number** (PIXIT) received. A 181 Being Forwarded is sent. The Redirection number included in the ACM is mapped into the History-Info header in the 181 Being Forwarded. A Privacy header field "history" is escaped in the URI identified the diverted to user. The redirection reason is mapped into the cause-param in of the hi-targeted-uri identifying the diverted-to user.	
SIP Parameter values:	181 Being Forwarded: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=**Status-Code**; ?Privacy=history; index=1.1	
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 181 Being Forwarded ← ← ACM 180 Ringing ← ← CPG(Alerting) 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter	**Derived value of parameter field**	**SIP component**	**Value**
Call diversion information			**History-Info header**
Redirection reason	**ISUP_REASON**	Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*BCI called party status "subscriber free" in ACM received, mapping of Redirection reason.* Ensure that the SUT on receipt of an **ACM** message indicating a first diversion with the **Backward call indicators parameter** coded Called party's status indicator = subscriber free the **Call diversion information parameter** coded Notification subscription option = "010"B Redirection reason = **ISUP_REASON** and the **Generic notification indicator parameter** coded Notification indicator = call is diverting, **Redirection number** (PIXIT) received A 180 Ringing is sent. The Redirection number included in the ACM is mapped into the History-Info header in the 180 Ringing. A Privacy header field "history" is escaped in the URI identified the diverted to user. The redirection reason is mapped into the cause-param in of the hi-targeted-uri identifying the diverted-to user.	
SIP Parameter values:	180 Ringing: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=**Status-Code**; ?Privacy=history; index=1.1	
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter	**Derived value of parameter field**	**SIP component**	**Value**
Call diversion information			**History-Info header**
Redirection reason	**ISUP_REASON**	Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509007	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*CPG with Event indicator ALERTING received, mapping of Redirection reason.* Ensure that the SUT, on receipt of a **CPG** message indicating a first diversion with the **Event information parameter** coded Event indicator = ALERTING, the **Call diversion information parameter** coded Notification subscription option = "010"B Redirection reason = **ISUP_REASON** and the **Generic notification indicator parameter** coded Notification indicator = call is diverting, **Redirection number** (PIXIT) received. A 180 Ringing is sent. The Redirection number included in the CPG is mapped into the History-Info header in the 180 Ringing. A Privacy header field "history" is escaped in the URI identified the diverted to user. The redirection reason is mapped into the cause-param in of the hi-targeted-uri identifying the diverted-to user.	
SIP Parameter values:	180 Ringing: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=**Status-Code**; ?Privacy=history; index=1.1	
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM ← ACM(no indication) 180 Ringing ← ← CPG 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter	**Derived value of parameter field**	**SIP component**	**Value**
Call diversion information			**History-Info header**
Redirection reason	**ISUP_REASON**	Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509008	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*CPG with Event indicator PROGRESS received, mapping of Redirection reason.* Ensure that the SUT, on receipt of a **CPG** message indicating a first diversion with the **Event information parameter** coded Event indicator = PROGRESS, the **Call diversion information parameter** coded Notification subscription option = "010"B Redirection reason = **ISUP_REASON** and the **Generic notification indicator parameter** coded Notification indicator = call is diverting, **Redirection number** (PIXIT) received A 181 Being Forwarded is sent. The Redirection number included in the CPG is mapped into the History-Info header in the 181 Being Forwarded. A Privacy header field "history" is escaped in the URI identified the diverted to user. The redirection reason is mapped into the cause-param in of the hi-targeted-uri identifying the diverted-to user.	
SIP Parameter values:	181 Being Forwarded: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=**Status-Code**; ?Privacy=history; index=1.1	
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 181 Being Forwarded ← ← CPG 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter	**Derived value of parameter field**	**SIP component**	**Value**
Call diversion information			**History-Info header**
Redirection reason	**ISUP_REASON**	Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509009	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*BCI called party status "no indication" in ACM received, mapping of Notification subscription option.* Ensure that the SUT, on receipt of an **ACM** message indicating a first diversion with the **Backward call indicators parameter** coded Called party's status indicator = no indication the **Call diversion information parameter** coded **Notification subscription option** = ISUP_NSO Redirection reason = unconditional and the **Generic notification indicator parameter** coded Notification indicator = call is diverting, **Redirection number** (PIXIT) received. A 181 Being Forwarded is sent. The Redirection number included in the ACM is mapped into the History-Info header in the 181 Being Forwarded. A Privacy header field "history" is escaped in the URI identified the diverted to user.	
SIP Parameter values:	181 Being Forwarded: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=302; ?priv-value; index=1.1	
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 181 Being Forwarded ← ← ACM 180 Ringing ← ← CPG(Alerting) 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

	SIP component **History-Info header,** **priv-value component**	**Call diversion information *Notification subscription options*** **ISUP_NSO**
VA_01	Privacy header field absent or "none"	ISUP_NSO = presentation allowed with redirection number
VA_02	Privacy "history"	ISUP_NSO = presentation allowed without redirection number

TP509010	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*BCI called party status "subscriber free" in ACM received, mapping of Notification subscription option.* Ensure that the SUT on receipt of an **ACM** message indicating a first diversion with the **Backward call indicators parameter** coded Called party's status indicator = subscriber free the **Call diversion information parameter** coded Notification subscription option = ISUP_NSO Redirection reason = unconditional and the **Generic notification indicator parameter** coded Notification indicator = call is diverting, **Redirection number** (PIXIT) received. A 180 Ringing is sent. The Redirection number included in the ACM is mapped into the History-Info header in the 180 Ringing. A Privacy header field "history" is escaped in the URI identified the diverted to user.	
SIP Parameter values:	180 Ringing: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=302; ?priv-value; index=1.1	
ISUP Parameter values:	ANM: Redirection number restriction	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

	SIP component History-Info header, priv-value component	**Call diversion information *Notification subscription options* ISUP_NSO**
VA_01	Privacy header field absent or "none"	ISUP_NSO = presentation allowed with redirection number
VA_02	Privacy "history"	ISUP_NSO = presentation allowed without redirection number

TP509011	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*CPG with Event indicator ALERTING received, mapping of Notification subscription option.* Ensure that the SUT on receipt of a **CPG** message indicating a first diversion with the **Event information parameter** coded Event indicator = ALERTING, the **Call diversion information parameter** coded Notification subscription option = ISUP_NSO Redirection reason = unconditional and the **Generic notification indicator parameter** coded Notification indicator = call is diverting, **Redirection number** (PIXIT) received. A 180 Ringing is sent. The Redirection number included in the CPG is mapped into the History-Info header in the 180 Ringing. A Privacy header field "history" is escaped in the URI identified the diverted to user.	
SIP Parameter values:	180 Ringing: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=302; ?priv-value; index=1.1	
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM ← ACM 180 Ringing ← ← CPG 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

	SIP component **History-Info header,** **priv-value component**	**Call diversion information *Notification subscription options*** **ISUP_NSO**
VA_01	Privacy header field absent or "none"	ISUP_NSO = presentation allowed with redirection number
VA_02	Privacy "history"	ISUP_NSO = presentation allowed without redirection number

TP509012	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*CPG with Event indicator ALERTING received, mapping of Redirection number restriction.* Ensure that the SUT on receipt of a **CPG** message indicating a first diversion with the **Event information parameter** coded Event indicator = ALERTING, Redirection number restriction parameter = ISUP_RDIR_RESTR A 180 Ringing including a History-Info header is sent. A Privacy header field "history" is escaped in the URI identified the diverted to user.	
SIP Parameter values:	180 Ringing: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=302; ?priv-value; index=1.1	
ISUP Parameter values:	ANM: Redirection number restriction	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 181 Being Forwarded ← ← ACM 180 Ringing ← ← CPG 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

	Derived escaped SIP priv-value component	**Derived value of parameter field**
VA_01	Privacy header field absent or "none"	ISUP_RDIR_RESTR = Presentation allowed, '00'B
VA_02	Privacy header field "history" and "id"	ISUP_RDIR_RESTR = presentation restricted, '01'B
VA_03	Privacy header field absent or "none"	ISUP_RDIR_RESTR absent

TP509013	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*ANM received, mapping of the Redirection number restriction parameter.* Ensure that the SUT, on receipt of an **ANM** message with the Redirection number restriction parameter = ISUP_RDIR_RESTR or parameter absent, a 200 IK INVITE including a History-Info header is sent. A Privacy header field is escaped in the URI identified the diverted to user according the value of the Redirection number restriction parameter.	
SIP Parameter values:	200 OK INVITE: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=302; ?priv-value; index=1.1	
ISUP Parameter values:	ANM: Redirection number restriction	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 181 Being Forwarded ← ← ACM 180 Ringing ← ← CPG 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

	Derived escaped SIP component	**Derived value of parameter field**
VA_01	Privacy header field absent or "none"	ISUP_RDIR_RESTR = Presentation allowed, '00'B
VA_02	Privacy header field "history" and "id"	ISUP_RDIR_RESTR = presentation restricted, '01'B
VA_03	Privacy header field absent or "none"	ISUP_RDIR_RESTR absent

TP509014	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 1/5 AND PICS 10/6	
Test purpose:	*CDIV performed, the first hi-targeted-to-uri is sent in the original called number national number* Ensure that the SUT in the Idle state Sends an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, Privacy header field is absent and with the complete **Original called number parameter** contained in the URI of first Index entry of History-Info header in the format "+" CC NDC SN. of a **IAM** message with the **Redirection information parameter** coded Redirection counter = 1 Redirecting reason = ISUP_RR and the **Original called number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals included in the format CC+NDC+SN.	
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=Status-Code ; index=1.1	
ISUP Parameter values:	IAM: **Original called number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals userinfo of the hi-targeted-to from index 1	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter	**Derived value of parameter field**	SIP component	**Value**
IAM		INVITE	
Redirection Information	**Redirecting reason** ISUP_RR	Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509015	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:	PICS 1/5 AND PICS 10/6	
Test purpose:	*CDIV performed, the first hi-targeted-to-uri is sent in the original called number international number* Ensure that the SUT in the Idle state, on receipt an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, the Privacy header is ansent and with the complete **Original called number parameter** contained URI of first Index entry of History-Info header in the format "+" CC NDC SN. Sends of a **IAM** message with the **Redirection information parameter** coded Redirection counter = 1 Redirecting reason = ISUP_RR and the **Original called number parameter** coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals included in the format CC+NDC+SN	
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=Status-Code; index=1.1	
ISUP Parameter values:	IAM: **Original called number parameter** coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals userinfo of the hi-targeted-to from index 1	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter or IE	**Derived value of parameter field**	SIP component	**Value**
IAM		INVITE	
Redirection Information	**Redirecting reason** ISUP_RR	Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509016	**SIP reference: RFC 3261 [6]**				**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /				
SIP selection criteria:					
ISUP selection criteria:	PICS 10/6				
Test purpose:	*CDIV performed, the first hi-targeted-to-uri is sent in the original called number Privacy header is equal "history"* Ensure that the SUT in the Idle state, INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, the priv-value set to "history" and with the complete **Original called number parameter** contained URI of first Index entry of History-Info header in the format "+" CC NDC SN. Sends an a **IAM** message with the **Redirection information parameter** coded Redirection counter = 1 Redirecting reason = ISUP_RR and the **Original called number parameter** coded Address presentation restricted parameter = presentation restricted				
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user?Privacy=history; index=1, hi-targeted-ti uri diverted to user; cause=Status-Code; index=1.1				
ISUP Parameter values:	IAM: **Original called number parameter** coded Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals userinfo of the hi-targeted-to from index 1				
Comments:	**SIP**		**SUT**		**ISUP**
	INVITE	→		→	IAM
	180 Ringing	←		←	ACM
	200 OK INVITE	←		←	ANM
	ACK	→			
			Communication		
	BYE	→		→	REL
	200 OK BYE	←		←	RLC

ISUP Parameter or IE	**Derived value of parameter field**	**SIP component**	**Value**
IAM		INVITE	
Redirection Information	**Redirecting reason** ISUP_RR	Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509017	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 1/5 AND PICS 10/6	
Test purpose:	*CDIV performed, the first hi-targeted-to-uri is sent in the redirecting number national number* Ensure that the SUT in the Idle state Sends an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, Privacy header field is absent and with the complete **Redirecting number parameter** contained in the hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN. of a **IAM** message with the **Redirection information parameter** coded Redirection counter = 1 Redirecting reason = ISUP_RR and the **Redirecting number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals included in the format CC+NDC+SN	
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=Status-Code; index=1.1	
ISUP Parameter values:	IAM: **Redirecting number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals userinfo of the hi-targeted-to from index 1	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter or IE	**Derived value of parameter field**	SIP component	**Value**
IAM		INVITE	
Redirection Information	**Redirecting reason** ISUP_RR	Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509018	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:	PICS 1/5 AND PICS 10/6	
Test purpose:	*CDIV performed, the first hi-targeted-to-uri is sent in the redirecting number international number* Ensure that the SUT in the Idle state, on receipt an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, Privacy header field is absent and with the complete **Redirecting number parameter** contained hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN. Sends of a **IAM** message with the **Redirection information parameter** coded Redirection counter = 1 Redirecting reason = ISUP_RR and the **Redirecting number parameter** coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals included in the format CC+NDC+SN	
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause=Status-Code; index=1.1	
ISUP Parameter values:	IAM: **Redirecting number parameter** coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals userinfo of the hi-targeted-to from index 1	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter or IE	**Derived value of parameter field**	SIP component	**Value**
IAM		INVITE	
Redirection Information	**Redirecting reason** ISUP_RR	Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509019	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:	PICS 10/6	
Test purpose:	*CDIV performed, the second hi-targeted-to-uri is sent in the redirecting number Privacy header is equal "history"* Ensure that the SUT in the Idle state, INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, the priv-value set to "history" and with the complete **Redirecting number parameter** contained hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN. Sends an a **IAM** message with the **Redirection information parameter** coded Redirection counter = 1 Redirecting reason = ISUP_RR and the **Redirecting number parameter** coded Address presentation restricted parameter = presentation restricted	
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user?Privacy=history; index=1, hi-targeted-ti uri diverted to user; cause=Status-Code; index=1.1	
ISUP Parameter values:	IAM: **Redirecting number parameter** coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals userinfo of the hi-targeted-to from index 1	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter or IE	**Derived value of parameter field**	SIP component	**Value**
IAM		INVITE	
Redirection Information	**Redirecting reason** ISUP_RR	Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509020	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:	PICS 10/	
Test purpose:	*CDIV performed, the second hi-targeted-to-uri Privacy header is not included is sent in the redirecting number and the first hi-targeted-to-uri Privacy header is sent in the original called number* Ensure that the SUT in the Idle state, on receipt of an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table and with the complete **Original called number parameter** contained the URI of first Index entry of History-Info header, the Privacy header field is absent in the format "+" CC NDC SN. The **Redirecting number parameter** is contained in the second hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN the Privacy header field is absent. Sends a **IAM** message with the **Redirection information parameter** coded Redirection counter 2 Redirecting reason = ISUP_RR, the **Original called number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals included and the **Redirecting number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals included	
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user C; cause=302; index=1.1 hi-targeted-ti uri diverted to user D; cause=Status-Code; index=1.1.1	
ISUP Parameter values:	IAM: **Original called number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals userinfo of the hi-targeted-to from index 1 **Redirecting number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals userinfo of the hi-targeted-to from index 1.1	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter or IE	**Derived value of parameter field**	SIP component	**Value**
IAM		INVITE	
Redirection Information	**Redirecting reason** ISUP_RR	Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509021	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:	PICS 10/6	
Test purpose:	*CDIV performed, the second hi-targeted-to-uri Privacy ="history" is sent in the redirecting number and the first hi-targeted-to-uri Privacy "header" is sent in the original called number* Ensure that the SUT in the Idle state, on receipt of an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table and with the complete **Original called number parameter** contained the URI of first Index entry of History-Info header, the Privacy header field is absent in the format "+" CC NDC SN. The **Redirecting number parameter** is contained in the second hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN, the Privacy value is set to "history". Sends a **IAM** message with the **Redirection information parameter** coded Redirection counter 2 Redirecting reason = ISUP_RR, the **Original called number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals included and the **Redirecting number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals included	
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user C?Privacy=history; cause=302; index=1.1 hi-targeted-ti uri diverted to user D; cause=Status-Code; index=1.1.1	
ISUP Parameter values:	IAM: **Original called number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals userinfo of the hi-targeted-to from index 1 **Redirecting number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals userinfo of the hi-targeted-to from index 1.1	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter or IE	**Derived value of parameter field**	SIP component	**Value**
IAM		INVITE	
Redirection Information		Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509022	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:	PICS 10/6	
Test purpose:	*CDIV performed, the second hi-targeted-to-uri Privacy header absent is sent in the redirecting number and the first hi-targeted-to-uri Privacy = "history" is sent in the original called number Privacy* Ensure that the SUT in the Idle state, on receipt of an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table and with the complete **Original called number parameter** contained the URI of first Index entry of History-Info header, the Privacy value set to "history" in the format "+" CC NDC SN. The **Redirecting number parameter** is contained in the second hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN the Privacy header field is absent. Sends an **IAM** message with the **Redirection information parameter** coded Redirection counter 2 Redirecting reason = ISUP_RR, the **Original called number parameter** coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals included and the **Redirecting number parameter** coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals included	
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user?Privacy=history; index=1, hi-targeted-ti uri diverted to user C; cause=302; index=1.1 hi-targeted-ti uri diverted to user D; cause=Status-Code; index=1.1.1	
ISUP Parameter values:	IAM: **Original called number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address userinfo of the hi-targeted-to from index 1 **Redirecting number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation allowed Address signals userinfo of the hi-targeted-to from index 1.1	

Comments:

SIP		**SUT**		**ISUP**
INVITE	→		→	IAM
180 Ringing	←		←	ACM
200 OK INVITE	←		←	ANM
ACK	→			
		Communication		
BYE	→		→	REL
200 OK BYE	←		←	RLC

ISUP Parameter or IE	**Derived value of parameter field**	SIP component	**Value**
IAM		INVITE	
Redirection Information		Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509023	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:	PICS 10/6	
Test purpose:	*CDIV performed, the second hi-targeted-to-uri Privacy="history" is sent in the redirecting number and the first hi-targeted-to-uri Privacy = "history" is sent in the original called number Privacy* Ensure that the SUT in the Idle state, on receipt of an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table and with the complete **Original called number parameter** contained the URI of first Index entry of History-Info header the Privacy is set to "history" in the format "+" CC NDC SN. The **Redirecting number parameter** is contained in the second hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN the Privacy is set to "history". Sends an **IAM** message with the **Redirection information parameter** coded Redirection counter 2 Redirecting reason = ISUP_RR, the **Original called number parameter** coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals included and the **Redirecting number parameter** coded Nature of address indicator = international number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals included	
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user?Privacy=history; index=1, hi-targeted-ti uri diverted to user C?Privacy=history; cause=302; index=1.1 hi-targeted-ti uri diverted to user D; cause=Status-Code; index=1.1.1	
ISUP Parameter values:	IAM: **Original called number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals userinfo of the hi-targeted-to from index 1 **Redirecting number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address presentation restricted parameter = presentation restricted Address signals userinfo of the hi-targeted-to from index 1.1	
Comments:	**SIP** **SUT** **ISUP** INVITE ➔ ➔ IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK ➔ **Communication** BYE ➔ ➔ REL 200 OK BYE ← ← RLC	

ISUP Parameter or IE	**Derived value of parameter field**	SIP component	**Value**
IAM		INVITE	
Redirection Information		Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509024	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*CDIV performed, the third hi-targeted-to-uri is sent in the redirecting Redirection counter is mapped from the latest history entry* Ensure that the SUT in the Idle state, on receipt of an INVITE message containing a History-Info header with three History entries, the hi-targeted-to-uri of first index is mapped into the **Original called number parameter**; the hi-targeted-to-uri of third index is mapped into the **Redirecting number parameter** Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table. Sends a **IAM** message with the **Redirection information parameter** coded Redirection counter 3 Redirecting reason = ISUP_RR, the **Original called number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address signals included and the **Redirecting number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address signals included	
SIP Parameter values:	INVITE: History-Info: hi-targeted-to uri served user; index=1, hi-targeted-ti uri diverted to user C; cause=302index=1.1 hi-targeted-ti uri diverted to user D; cause=486index=1.1.1 hi-targeted-ti uri diverted to user E; cause=Status-Code; index=1.1.1.1	
ISUP Parameter values:	IAM: **Original called number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address signals userinfo of the hi-targeted-to from index 1 **Redirecting number parameter** coded Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony numbering plan Address signals userinfo of the hi-targeted-to from index 1.1.1	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

ISUP Parameter or IE	**Derived value of parameter field**	SIP component	**Value**
IAM		INVITE	
Redirection Information		Cause Value in History Index; cause-param = "cause" EQUAL **Status-Code**	**Cause value**
	unknown '0000'B		404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP509025	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	SIP-ISUP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 5/12 AND NOT PICS 5/13 AND NOT PICS 5/14 AND NOT PICS 5/15	
Test purpose:	*Interworking not supported, session successful.* Ensure that the SUT in the Idle state, on receipt of an INVITE message containing a History-Info header with three History entries the History-Info header entries are not mapped into any call diversion relateded parameters in the IAM and the session setup is not disruppted.	
SIP Parameter values:	INVITE: History-Info: hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user C; cause=Status code; index=1.1 hi-targeted-ti uri diverted to user D; cause=Status code; index=1.1.1	
ISUP Parameter values:	IAM: no mapping	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM 200 OK INVITE ← ← ANM ACK → **Communication** BYE → → REL 200 OK BYE ← ← RLC	

6.3.1.10 Call waiting (CW)

TP510001	**SIP reference: RFC 3261 [6]**				**ISUP reference: ES 283 027 [1], clause 7.4.9**
TSS reference:	SIP-ISUP/SS/CW/				
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	*No mapping of Call Waiting indication in the ACM at the I-MGCF* Ensure that the indication for Call Waiting contained in an ACM, Generic notification "call is a waiting call" is not interworked in a 180 Ringing Response				
SIP Parameter values:					
ISUP Parameter values:	ACM: Generic notification parameter = "Call is a waiting call"				
Comments:	**SIP**		**SUT**		**ISUP**
	INVITE	→		→	IAM
	180 Ringing	←		←	ACM
			Ringing tone		
	200 OK INVITE	←		←	ANM
	ACK	→			
			Conversation		
	BYE	←		←	REL
	200 OK BYE	→		→	RLC

TP510002	**SIP reference: RFC 3261 [6]**				**ISUP reference: ES 283 027 [1], clause 7.4.9**
TSS reference:	SIP-ISUP/SS/CW/				
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	*No mapping of Call Waiting indication in the CPG at the I-MGCF* Ensure that the indication for Call Waiting contained in an CPG, Generic notification "call is a waiting call", is not interworked in a 180 Ringing Response				
SIP Parameter values:	180 Ringing				
ISUP Parameter values:	ACM: Called party status "no indication" CPG: Generic notification parameter = "Call is a waiting call"				
Comments:	**SIP**		**SUT**		**ISUP**
	INVITE	→		→	IAM
				←	ACM
	180 Ringing	←		←	CPG
			Ringing tone		
	200 OK INVITE	←		←	ANM
	ACK	→			
			Conversation		
	BYE	←		←	REL
	200 OK BYE	→		→	RLC

6.3.1.11 User to user signalling (UUS)

TP511001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33], clause 1.3.7.2**
TSS reference:	SIP-ISUP/SS/UUS/	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	Explicit request supported, a *FAR user-to-user service 3 request (not essential) is rejected with FRJ* Ensure that the SUT if a FAR is received with an **user-to-user service 3 request (not essential)** after call setup, sent a FRJ to reject the request. The SIP signalling procedure is not disrupted.	
SIP Parameter values:		
ISUP Parameter values:	FRJ: User-to-user indicator = "Service 3 not provided"	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation ← FAR → FRJ Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP511002	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737 [33], clause 1.3.5.2.5.2**
TSS reference:	SIP-ISUP/SS/UUS/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	*Explicit request not supported, no response on receipt of a FAR* Ensure that the SUT if a FAR is received with an user-to-**user service 3 request (not essential)** after call setup, the SIP signalling procedure is not disrupted.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation ← FAR Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP511003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33],** **clause 1.3.5.2.5.2**
TSS reference:	SIP-ISUP/SS/UUS/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*User-to-user service 1 implicit request* Ensure that the SUT if an User-to-User header is included in the INVITE a User-to-user parameter is sent in the IAM. The data field of the User-to-user information is derived from the uuidata component of the User-to-User header in the INVITE	
SIP Parameter values:	INVITE: User-to-User uuidata	
ISUP Parameter values:	IAM: User-to-user information	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP511004	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737 [33],** **clause 1.3.5.2.5.2.1**
TSS reference:	SIP-ISUP/SS/UUS/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*User-to-user service 1 response* Ensure that the User-to-user information parameter in an ACM, ANM or REL is mapped in uuidata of a nUser-to-User header sent in an Provisional response, final response or BYE	
SIP Parameter values:	18x: User-to-User uuidata 200: User-to-User uuidata BYE: User-to-User uuidata	
ISUP Parameter values:	ACM: User-to-user information ANM: User-to-user information REL: User-to-user information	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

6.3.1.12 Explicit call transfer (ECT)

TP512001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.8**
TSS reference:	SIP-ISUP/SS/ECT/	
SIP selection criteria:		
ISUP selection criteria:	PICS 12/1	
Test purpose:	*Loop prevention procedure supported, a LOP response "insufficient information" is sent* Ensure that the SUT if a LOP(request) is received returns a LOP (response) with the indication "insufficient information" continue without disrupting the SIP signalling procedure. Ensure that the SUT if a FAC is received continue without disrupting the SIP signalling procedure.	
SIP Parameter values:		
ISUP Parameter values:	LOP: Response "insufficient information"	
Comments:		

SIP		**SUT**		**ISUP**
INVITE	→		→	IAM
180 Ringing	←		←	ACM
		Ringing tone		
200 OK INVITE	←		←	ANM
ACK	→			
		Conversation		
			←	LOP
			→	LOP
			←	FAC
		Conversation		
BYE	←		←	REL
200 OK BYE	→		→	RLC

TP512002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.8**
TSS reference:	SIP-ISUP/SS/ECT/	
SIP selection criteria:		
ISUP selection criteria:	NO PICS 12/1	
Test purpose:	*Loop prevention procedure not supported* Ensure that the SUT if a LOP(request) is received continue without disrupting the SIP signalling procedure. Ensure that the SUT if a FAC is received continue without disrupting the SIP signalling procedure.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation ← LOP ← FAC Conversation BYE ← ← REL 200 OK BYE → → RLC	

6.3.1.13 Completion of Call to Busy Subscriber (CCBS)

TP513001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.11**
TSS reference:	SIP-ISUP/SS/CCBS/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CCBS possible in the Diagnostics field received* Ensure that the SUT if a REL is received contained a Diagnostic field and the CCBS indicator is coded as CCBS possible: • continue without disrupting the SIP signalling procedure.	
SIP Parameter values:		
ISUP Parameter values:	REL: Cause indicator Diagnostics CCBS possible	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 486 Busy Here ← ← REL ACK → → RLC	

6.3.1.14 Completion of Calls on No reply (CCNR)

TP514001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.12**
TSS reference:	SIP-ISUP/SS/CCNR/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*CCNR possible in the ACM received* Ensure that the SUT if a ACM is received and a CCNR Possible Indicator is included: • continue without disrupting the SIP signalling procedure.	
SIP Parameter values:		
ISUP Parameter values:	ACM: CCNR possible indicator CCNR possible	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

6.3.1.15 Anonymous Call Rejection (ACR)

TP515001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.23**
TSS reference:	SIP-ISUP/SS/ACR/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Mapping of cause value #24* Ensure that the SUT, if a destination user has subscribed the ACR supplementary service: • the call attempt is rejected with a REL cause value 24 "call rejected due to ACR supplementary service".	
SIP Parameter values:	INVITE: Privacy-header = "id" 603 Decline: Reason header field Reason: ITU-T Recommendation Q.850 [5]; cause=24	
ISUP Parameter values:	REL: **Cause value:** 24 "call rejected due to ACR supplementary service"	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 603 Decline ← ← REL ACK → → RLC	

6.3.1.16 Closed user group (CUG)

<table>
<tr><th>TP516001</th><th>SIP reference: RFC 3261 [6]</th><th colspan="3">ISUP reference:
7.4.1.1</th></tr>
<tr><td>TSS reference:</td><td colspan="4">SIP-ISUP/SS/CUG/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="4"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="4">PICS 5/7</td></tr>
<tr><td>Test purpose:</td><td colspan="4"><i>Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "00"</i>

Ensure that the <cugCommunicationIndicator> value "00" contained in the INVITE <cug> XML body is sent in a optional forward call indicator - CUG call indicator, if any other value of the optional forward call indicator have to be set not equal "0". No mapping of <networkIndicator> and <cugInterlockBinaryCode> into Closed User Group interlock code.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="4">INVITE:
<cug>
<networkIndicator>[PIXIT]</networkIndicator>
<cugInterlockBinaryCode>[PIXIT]</cugInterlockBinaryCode>
<cugCommunicationIndicator>00</cugCommunicationIndicator>
</cug></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="4">IAM:
Optional Forward Call Indicator CUG call indicator = "00"
When optional forward call indicator have to be sent in case of an other indicator is not set to "0"</td></tr>
<tr><td rowspan="9">Comments:</td><td>SIP</td><td></td><td>SUT</td><td>ISUP</td></tr>
<tr><td>INVITE</td><td>→</td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td>←</td><td>ACM</td></tr>
<tr><td></td><td colspan="2">Ringing tone</td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td></tr>
<tr><td></td><td colspan="2">Conversation</td><td></td></tr>
<tr><td>BYE</td><td>←</td><td>←</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td>→</td><td>RLC</td></tr>
</table>

<table>
<tr><td>TP516002</td><td>SIP reference: RFC 3261 [6]</td><td>ISUP reference: 7.4.1.1</td></tr>
<tr><td>TSS reference:</td><td colspan="2">SIP-ISUP/SS/CUG/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 5/7</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "01"</i>

Ensure that the <cugCommunicationIndicator> value "01" contained in the INVITE <cug> XML body is not sent in a optional forward call indicator - CUG call indicator. If the optional forward call indicator have to be sent, the CUG call indicator is set to "00" no CUG call. No mapping of <networkIndicator> and <cugInterlockBinaryCode> into Closed User Group interlock code.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INVITE:
<cug>
    <networkIndicator>[PIXIT]</networkIndicator>
    <cugInterlockBinaryCode>[PIXIT]</cugInterlockBinaryCode>
    <cugCommunicationIndicator>01</cugCommunicationIndicator>
</cug></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM:
Optional Forward Call Indicator CUG call indicator = "00"
When optional forward call indicator have to be sent in case of an other indicator is not set to "0"</td></tr>
<tr><td>Comments:</td><td colspan="2"><table>
<tr><td>SIP</td><td></td><td>SUT</td><td></td><td>ISUP</td></tr>
<tr><td>INVITE</td><td>→</td><td></td><td>→</td><td>IAM</td></tr>
<tr><td>180 Ringing</td><td>←</td><td></td><td>←</td><td>ACM</td></tr>
<tr><td></td><td></td><td>Ringing tone</td><td></td><td></td></tr>
<tr><td>200 OK INVITE</td><td>←</td><td></td><td>←</td><td>ANM</td></tr>
<tr><td>ACK</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>BYE</td><td>←</td><td></td><td>←</td><td>REL</td></tr>
<tr><td>200 OK BYE</td><td>→</td><td></td><td>→</td><td>RLC</td></tr>
</table></td></tr>
</table>

TP516003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.1.1**
TSS reference:	SIP-ISUP/SS/CUG/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/7	
Test purpose:	*Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "10"* Ensure that the <cugCommunicationIndicator> value "10" contained in the INVITE <cug> XML body is sent in a optional forward call indicator - CUG call indicator ="10". The XML <cug> <**networkIndicator**> is mapped into the IAM Closed User Group interlock code **Network indentity** and the XML <cug> <**cugInterlockBinaryCode**> is mapped into the IAM Closed User Group interlock code **Binary code**.	
SIP Parameter values:	INVITE: <cug> <networkIndicator>[PIXIT]</networkIndicator> <cugInterlockBinaryCode>[PIXIT]</cugInterlockBinaryCode> <cugCommunicationIndicator>10</cugCommunicationIndicator> </cug>	
ISUP Parameter values:	IAM: Optional Forward Call Indicator CUG call indicator = "10" Closed User Group interlock code **Binary code** derived from INVITE XML body <**cugInterlockBinaryCode**> **Network indentity** derived from INVITE XML body <**networkIndicator**>	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP516004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.1.1**
TSS reference:	SIP-ISUP/SS/CUG/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/7	
Test purpose:	*Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "11"* Ensure that the <cugCommunicationIndicator> value "11" contained in the INVITE <cug> XML body is sent in a optional forward call indicator - CUG call indicator ="11". The XML <cug> <**networkIndicator**> is mapped into the IAM Closed User Group interlock code **Network indentity** and the XML <cug> <**cugInterlockBinaryCode**> is mapped into the IAM Closed User Group interlock code **Binary code**.	
SIP Parameter values:	INVITE: <cug> <networkIndicator>[PIXIT]</networkIndicator> <cugInterlockBinaryCode>[PIXIT]</cugInterlockBinaryCode> <cugCommunicationIndicator>11</cugCommunicationIndicator> </cug>	
ISUP Parameter values:	IAM: Optional Forward Call Indicator CUG call indicator = "11" Closed User Group interlock code **Binary code** derived from INVITE XML body <**cugInterlockBinaryCode**> **Network indentity** derived from INVITE XML body <**networkIndicator**>	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP516005	**SIP reference: RFC 3261 [6]**	**ISUP reference: 7.4.1.1**
TSS reference:	SIP-ISUP/SS/CUG/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 5/7	
Test purpose:	*Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "10". The PSTN/ISDN network does not support CUG.* Ensure that the <cugCommunicationIndicator> value "10" contained in the INVITE <cug> XML body is not sent in a optional forward call indicator - CUG call indicator ="10" when the PSTN/ISDN does not support CUG. If the optional forward call indicator have to be sent, the CUG call indicator is set to "00" no CUG call. No mapping of <networkIndicator> and <cugInterlockBinaryCode> into Closed User Group interlock code.	
SIP Parameter values:	INVITE: <cug> <networkIndicator>[PIXIT]</networkIndicator> <cugInterlockBinaryCode>[PIXIT]</cugInterlockBinaryCode> <cugCommunicationIndicator>10</cugCommunicationIndicator> </cug>	
ISUP Parameter values:	IAM: Optional Forward Call Indicator CUG call indicator = "00" If optional forward call indicator have to be sent in case of an other indicator is not set to "0"	
Comments:	**SIP** **SUT** **ISUP** INVITE → → IAM 180 Ringing ← ← ACM Ringing tone 200 OK INVITE ← ← ANM ACK → Conversation BYE ← ← REL 200 OK BYE → → RLC	

TP516006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.1.1**
TSS reference:	SIP-ISUP/SS/CUG/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 5/7	
Test purpose:	*Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "11". The PSTN/ISDN network does not support CUG.* Ensure that the <cugCommunicationIndicator> value "11" contained in the INVITE <cug> XML body is sent in a optional forward call indicator - CUG call indicator ="11". The XML <cug> <**networkIndicator**> is mapped into the IAM Closed User Group interlock code **Network indentity** and the XML <cug> <**cugInterlockBinaryCode**> is mapped into the IAM Closed User Group interlock code **Binary code**.	
SIP Parameter values:	INVITE: <cug> <networkIndicator>[PIXIT]</networkIndicator> <cugInterlockBinaryCode>[PIXIT]</cugInterlockBinaryCode> <cugCommunicationIndicator>11</cugCommunicationIndicator> </cug>	
ISUP Parameter values:		
Comments:	**SIP** **SUT** **ISUP** INVITE → 403 Forbidden ← ACK →	

6.3.2 Interworking from ISUP to SIP (Outgoing Call)

6.3.2.1 Calling Line Identification (CLI)

TP601001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*No calling party number and no additional calling party number received* Ensure that when the SUT has received an IAM message whereby Calling Party Number parameter and the Generic Number are not contained: • Sends an INVITE message without the "P-Asserted-Identity header field", a "From header field" set to unavailable@anonymous.invalid and without a Privacy Header field.	
SIP Parameter values:	INVITE: From <unavailable@anonymous>	
ISUP Parameter values:	IAM: no Calling party number no Generic Number: "Additional calling party number"	
Comments:		

ISUP/BICC		**SUT**		**SIP**
IAM	→		→	INVITE
ACM	←		←	180 Ringing
	Ringing tone			
ANM	←		←	200 OK INVITE
			→	ACK
		Conversation		
REL	→		→	BYE
RLC	←		←	200 OK BYE

TP601002	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*No calling party number and additional calling party number received* Ensure that when the SUT has received an IAM message whereby Calling Party Number parameter is **not** contained and the Generic Number is contained whereby the address presentation restriction parameter is set to "presentation allowed" and the Nature of Address Indicator is set to NoAS_VALUE: • Sends an INVITE message without the "P-Asserted-Identity header field", a "From header field" and no Privacy Header field.	
SIP Parameter values:	P-Asserted-Identity header field: not included: From header field: Tel or SIP URI: Addr_SPEC_ID Derived from Generic Number parameter Address Signals (AcgPN) Privacy header: is not included	
ISUP Parameter values:	IAM: Generic Number: "additional calling party number" Nature of Address Indicator: NoAS_VALUE APRI "presentation allowed"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP601003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*No calling party number and additional calling party number presentation restricted received* Ensure that when the SUT has received an IAM message whereby Calling Party Number parameter is **not** contained and the Generic Number is contained whereby the address presentation restriction parameter is set to "presentation restricted" and the Nature of Address Indicator is set to NoAS_VALUE: • Sends an INVITE message without the P-Asserted-Identity header field, a From header field set to unavailable@anonymous.invalid and no Privacy Header field.	
SIP Parameter values:	INVITE: From <unavailable@anonymous> P-Asserted-Identity header field: not included: Privacy header: is not included	
ISUP Parameter values:	IAM: Generic Number: "additional calling party number", Nature of Address Indicator: NoAS_VALUE APRI "presentation restricted"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Table 39

Values for test purpose TP601003		
	ISUP Parameter values:	**SIP Parameter values:**
VA_01	**IAM** NoAS_VALUE: "*national (significant) number"(NDC+SN)*	**INVITE** FHf_Addr_SPEC_ID: CC (of the country where the MGCF is located) is added to the Generic Number Address Signals and then mapped to user portion of URI scheme
VA_02	**IAM** NoAS_VALUE: "*international number" ("+"CC+NDC+SN)*	**INVITE** FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used.

TP601004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Calling party number presentation allowed and no additional calling party number received* Ensure that when the SUT has received an IAM message, the Calling Party Number is contained whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to "presentation allowed" and the Generic Number is not contained: • Sends an INVITE message with the "P-Asserted-Identity header field" where the „Tel or SIP URI" is set to PAIh_Addr_SPEC_ID, a "From header field" where the „Tel or SIP URI" is set to FHf_Addr_SPEC_ID without "Privacy Header field" or "id" is not included.	
SIP Parameter values:	P-Asserted-Identity header field: Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) From header field: Tel or SIP URI: Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) Privacy header: is not included or if included, "id" is not included	
ISUP Parameter values:	IAM: Calling party number APRI "presentation allowed" no Generic Number: "Additional calling party number"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Table 40

Values for test purpose TP601004		
	ISUP Parameter values:	**SIP Parameter values:**
VA_01	**IAM** NoAS_VALUE: "*national (significant) number"(NDC+SN)*	**INVITE** PAIh_Addr_SPEC_ID = FHf_Addr_SPEC_ID: Add CC (of the country where the MGCF is located) to CgPN Signals then map to user portion of URI scheme used
VA_02	**IAM** NoAS_VALUE: "*international number"* *("+"CC+NDC+SN)*	**INVITE** PAIh_Addr_SPEC_ID= FHf_Addr_SPEC_ID: the complete to CgPN Signals is mapped to the user portion of URI scheme.

TP601005	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Calling party number presentation restricted and no additional calling party number received* Ensure that when the SUT has received an IAM message, the Calling Party Number is contained whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to "presentation restricted" and the Generic Number is not contained: • Sends an INVITE message with the "P-Asserted-Identity header field" where the „Tel or SIP URI" is set to PAIh_Addr_SPEC_ID, a "From header field" set to unavailable@anonymous.invalid and with Privacy Header field value "id".	
SIP Parameter values:	P-Asserted-Identity header field: Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) From header field: Tel or SIP URI Tel or SIP URI: unavailable@anonymous.invalid Privacy header: "id".	
ISUP Parameter values:	IAM: Calling party number APRI "presentation restricted" no Generic Number: "Additional calling party number"	
Comments:	**ISUP/BICC** → **SUT** → **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP601006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Calling party number presentation restricted by the network and no additional calling party number received* Ensure that when the SUT has received an IAM message, the Calling Party Number is contained whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to "presentation restricted by the network" and the Generic Number is not contained: • Sends an INVITE message with the "P-Asserted-Identity header field" where the „Tel or SIP URI" is set to PAIh_Addr_SPEC_ID, a "From header field" set to unavailable@hostportion and with Privacy Header field value "id".	
SIP Parameter values:	P-Asserted-Identity header field: Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) From header field: Tel or SIP URI Tel or SIP URI: unavailable@hostportion Privacy header: "id".	
ISUP Parameter values:	IAM: Calling party number APRI "presentation restricted by the network" no Generic Number: "Additional calling party number"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Table 41

Values for test purpose TP601006		
	ISUP Parameter values:	**SIP Parameter values:**
VA_01	**IAM** NoAS_VALUE: "*national (significant) number"(NDC+SN)*	**INVITE** PAIh_Addr_SPEC_ID = FHf_Addr_SPEC_ID: CC (of the country where the MGCF is located) is added to the CgPN Signals and then mapped to user portion of URI scheme used
VA_02	**IAM** NoAS_VALUE: "*international number" ("+"CC+NDC+SN)*	**INVITE** PAIh_Addr_SPEC_ID= FHf_Addr_SPEC_ID: the complete to CgPN Signals is mapped to the user portion of URI scheme.

TP601007	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Calling party number presentation allowed and an additional calling party number received* Ensure that when the SUT has received an IAM message, the Calling Party Number is contained whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to "presentation allowed" and the Generic Number is contained: • Sends an INVITE message with the "P-Asserted-Identity header field", where the „Tel or SIP URI" is set to PAIh_Addr_SPEC_ID "From header field" where the „Tel or SIP URI" is set to FH_Addr_SPEC_ID and without "Privacy Header field" or "id" is not included.	
SIP Parameter values:	P-Asserted-Identity header field: Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) From header field: Tel or SIP URI Tel or SIP URI: FH_Addr_SPEC_ID (Derived from Generic Number parameter Address Signals (AcgPN)) Privacy header: is not included or if included, "id" is not included.	
ISUP Parameter hbvalues:	IAM: Calling party number APRI "presentation allowed" Generic Number: "additional calling party number" Nature of Address Indicator: CP_NoAS_VALUE	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM ➔ ➔ INVITE ACM 🡐 🡐 180 Ringing Ringing tone ANM 🡐 🡐 200 OK INVITE ➔ ACK Conversation REL ➔ ➔ BYE RLC 🡐 🡐 200 OK BYE	

TP601008	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.2.3.1.2.6**
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*Calling party number presentation restricted and an additional calling party number received* Ensure that when the SUT has received an IAM message, the **Calling Party Number is** contained whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to **presentation restricted** and the **Generic Number** presentation allowed is included Sends an INVITE message with the "**P-Asserted-Identity header field** , where the "addr-spec" is set to PAIh_Addr_SPEC_ID "**From header field**" where the "addr-spec" is set to FH_Addr_SPEC_ID and with "Privacy Header field =id".	
SIP Parameter values:	P-Asserted-Identity header field: Tel or SIP URI: PAIh_Addr_SPEC_ID (**Derived from Calling Party Number parameter Address Signals**) From header field: addr-spec Tel or SIP URI: FH_Addr_SPEC_ID **(Derived from Generic Number parameter Address Signals (AcgPN))** Privacy header: "id"	
ISUP Parameter values:	IAM: Calling party number APRI "presentation restricted" Generic Number: "*additional calling party number"* Nature of Address Indicator: CP_NoAS_VALUE APRI: presentation restricted	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

Table 42

Values for test purpose TP601007; TP601008			
Test purposes	**ISUP Parameter values:**	**SIP Parameter values:**	
VA_01	**IAM** NoAS_VALUE: "*national (significant) number"(NDC+SN)*	**INVITE** FHf_Addr_SPEC_ID: Add CC (of the country where the MGCF is located) to Additional calling party number Signals then map to user portion of URI scheme used	**INVITE** PAIh_Addr_SPEC_ID: Add CC (of the country where the MGCF is located) to Calling party number Signals then map to user portion of URI scheme used
VA_02	**IAM** NoAS_VALUE: "*international number"* *("+"CC+NDC+SN)*	**INVITE** FHf_Addr_SPEC_ID: the complete Additional calling party number Address Signals is mapped to the user portion of URI scheme.	**INVITE** PAIh_Addr_SPEC_ID: the complete Calling party number Address Signals is mapped to the user portion of URI scheme used.

6.3.2.2 Call Hold (HOLD)

<table>
<tr><th>TP602001</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
7.4.10/[14]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP/SS/HOLD/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 8/4</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5">PICS 5/22</td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>Each party can hold and retrieve the remote party in the confirmed state</i>

Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold.

The calling party should be able to put the other party on hold
The calling party should be able to retrieve the other party
The called party should be able to put the other party on hold
The called party should be able to retrieve the other party</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">SDP: a=sendonly (put on hold)
a=sendrecv or omitted (retrieve the call)
o= . . <version incremented></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5">CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold)
Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)</td></tr>
<tr><td rowspan="12">Comments:</td><td>ISUP/BICC</td><td></td><td>MGCF</td><td></td><td>SIP</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td>CPG(hold)</td><td>→</td><td></td><td>→</td><td>INVITE(sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(recvonly)</td></tr>
<tr><td>CPG(retrieve)</td><td>→</td><td></td><td>→</td><td>INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(sendrecv)</td></tr>
<tr><td>CPG(hold)</td><td>←</td><td></td><td>←</td><td>INVITE(sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INVITE(recvonly)</td></tr>
<tr><td>CPG(retrieve)</td><td>←</td><td></td><td>←</td><td>INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INVITE(sendrecv)</td></tr>
</table>

TP602002	**SIP reference: RFC 3261 [6]**	**ISUP reference: 7.4.10/[14]**
TSS reference:	ISUP-SIP/SS/HOLD/	
SIP selection criteria:	PICS 8/4 AND PICS 8/1	
ISUP selection criteria:	PICS 5/22	
Test purpose:	*The calling party can hold and retrieve the remote party in the early dialogue* Ensure that a party can put the other party on hold in the alerting state. Ensure that the party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)	
Comments:	**ISUP/BICC** **MGCF** **SIP** IAM ➔ ➔ INVITE ACM ← ← 180 Ringing CPG(hold) ➔ ➔ UPDATE(sendonly) ← 200 OK UPDATE(recevonly) CPG(retrieve) ➔ ➔ UPDATE(sendrecv) ← 200 OK UPDATE(sendrecv)	

TP602003	**SIP reference: RFC 3261 [6]**	**ISUP reference: 7.4.10/[14]**
TSS reference:	ISUP-SIP/SS/HOLD/	
SIP selection criteria:	PICS 8/4	
ISUP selection criteria:	PICS 5/22	
Test purpose:	*HOLD indication in SDP in an UPDATE received* Ensure that a party can put the other party on hold after the calling user has provided all of the information necessary for processing the call. Ensure that the party can retrieve the call previously put on hold. The calling party should be able to put the other party on hold The calling party should be able to retrieve the other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . <version incremented>	
ISUP Parameter values:	ACM: called party status: no indication CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)	
Comments:	**ISUP/BICC** **MGCF** **SIP** IAM ➔ ➔ INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE CPG(hold) ← ← UPDATE(sendonly) ➔ 200 OK UPDATE(recevonly) CPG(retrieve) ← ← UPDATE(sendrecv) ➔ 200 OK UPDATE(sendrecv)	

<table>
<tr><th>TP602004</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
7.4.10/[14]</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/SS/HOLD/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 8/4 AND PICS 8/3</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 5/22</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>The SUT uses the UPDATE method to indicate HOLD in the SDP</i>

Ensure that a party can put the other party on hold in the alerting state. Ensure that the party can retrieve the call previously put on hold.

The calling party should be able to put the other party on hold
The calling party should be able to retrieve the other party</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">SDP: a=sendonly (put on hold)
a=sendrecv or omitted (retrieve the call)
o= . . <version incremented></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold)
Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>ISUP/BICC</th><th></th><th>MGCF</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td>CPG(hold)</td><td>→</td><td></td><td>→</td><td>UPDATE(sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE(recevonly)</td></tr>
<tr><td>CPG(retrieve)</td><td>→</td><td></td><td>→</td><td>UPDATE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK UPDATE(sendrecv)</td></tr>
</table>
</td></tr>
</table>

<table>
<tr><th>TP602005</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
7.4.10/[14]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP/SS/HOLD/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 8/4</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5">PICS 5/22</td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>Each party can hold and retrieve the remote party in the confirmed state</i>

Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold.

The calling party should be able to put the other party on hold
The called party should be able to put the other party on hold
The calling party should be able to retrieve the other party
The called party should be able to retrieve the other party</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">SDP: a=sendonly (put on hold)
a=sendrecv or omitted (retrieve the call)
o= . . <version incremented></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5">CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold)
Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)</td></tr>
<tr><td rowspan="12">Comments:</td><td>ISUP/BICC</td><td></td><td>MGCF</td><td></td><td>SIP</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td>CPG(hold)</td><td>→</td><td></td><td>→</td><td>INVITE(sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(recvonly)</td></tr>
<tr><td>CPG(hold)</td><td>←</td><td></td><td>←</td><td>INVITE(inactive)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INVITE(inactive)</td></tr>
<tr><td>CPG(retrieve)</td><td>→</td><td></td><td>→</td><td>INVITE(recvonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(sendonly)</td></tr>
<tr><td>CPG(retrieve)</td><td>←</td><td></td><td>←</td><td>INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INVITE(sendrecv)</td></tr>
</table>

<table>
<tr><th>TP602006</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
7.4.10/[14]</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP/SS/HOLD/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 8/4</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5">PICS 5/22</td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>Each party can hold and retrieve the remote party in the confirmed state</i>

Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold.

The calling party should be able to put the other party on hold
The called party should be able to put the other party on hold
The called party should be able to retrieve the other party
The calling party should be able to retrieve the other party</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">SDP: a=sendonly (put on hold)
a=sendrecv or omitted (retrieve the call)
o= . . <version incremented></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5">CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold)
Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)</td></tr>
<tr><td rowspan="12">Comments:</td><td>ISUP/BICC</td><td></td><td>MGCF</td><td></td><td>SIP</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td>CPG(hold)</td><td>→</td><td></td><td>→</td><td>INVITE(sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(recvonly)</td></tr>
<tr><td>CPG(hold)</td><td>←</td><td></td><td>←</td><td>INVITE(inactive)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INVITE(inactive)</td></tr>
<tr><td>CPG(retrieve)</td><td>←</td><td></td><td>←</td><td>INVITE(recvonly)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INVITE(sendonly)</td></tr>
<tr><td>CPG(retrieve)</td><td>→</td><td></td><td>→</td><td>INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(sendrecv)</td></tr>
</table>

6.3.2.3 Terminal portability (TP)

Void.

6.3.2.4 Conference calling (CONF)

TP604001	**SIP reference: RFC 3261 [6]**	**NGN reference:** **ES 283 027 [1], clause 7.4.14**
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:	PICS 8/2	
ISUP selection criteria:	PICS 5/10	
Test purpose:	*Establish and disconnect a Conference* Ensure that the SUT does not stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value **GEN_NOT_VALUE** was received due to the CONF supplementary service.	
SIP Parameter values:		
ISUP Parameter values:	CPG: Generic notification = **GEN_NOT_VALUE**	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation CPG(Conference established) → CPG(Conference disconnected) → Conversation REL → → BYE RLC ← ← 200 OK BYE	

Table 43

Values for test purpose TP604001, TP604002		
CPG→ **Generic notification** **GEN_NOT_VALUE**		
VA_01	Conference established	
VA_02	Conference disconnected	

<table>
<tr><th>TP604002</th><th>SIP reference: RFC 3261 [6]</th><th>NGN reference:
ES 283 027 [1], clause 7.4.14</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/SS/CONF/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 8/2</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 5/10</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Isolate and reattach a Conference</i>

Ensure that the SUT stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value GEN_NOT_VALUE was received due to the CONF supplementary service.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">SDP: a= a_LINE_VA (see table 43) or a line is omitted</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">CPG: Generic notification = Conference established
CPG: Generic notification = GEN_NOT_VALUE</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>ISUP/BICC</th><th></th><th>SUT</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td>Ringing tone</td><td></td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>CPG(Conference established)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>CPG(Isolated)</td><td>→</td><td></td><td>→</td><td>INVITE(sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(recvonly)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>CPG(Reattached)</td><td>→</td><td></td><td>→</td><td>INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>CPG(Conference disconnected)</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>
</td></tr>
</table>

Table 44

<table>
<tr><th colspan="3">Values for test purpose TP604001, TP604002</th></tr>
<tr><th colspan="2">CPG→
Generic notification
GEN_NOT_VALUE</th><th>INVITE/UPDATE→
SDP attribute line
a_LINE_VA</th></tr>
<tr><td>VA_01</td><td>isolated</td><td>sendonly or inactive</td></tr>
<tr><td>VA_02</td><td>reattached</td><td>sendrecv or recvonly or omitted</td></tr>
</table>

TP604003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.14** **ITU-T Rec Q.734.1 [34], clause 1.7**
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 5/10	
Test purpose:	*Mapping of isolated and reattached not supported* Ensure that the MGCF can receive in a CPG the Generic notifications "isolated" and "reattached", no mapping on the SIP side and the call is not disrupted. No mapping, no disrupting the SIP procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	CPG: Generic notification = Conference established CPG: Generic notification = isolated CPG: Generic notification = reattached CPG: Generic notification = Conference disconnected	
Comments:	see message flow below	

ISUP/BICC		**SUT**		**SIP**
IAM	→		→	INVITE
ACM	←		←	180 Ringing
Ringing tone				
ANM	←		←	200 OK INVITE
			→	ACK
		Conversation		
CPG(Conference established)	→			
CPG(Isolated)	→			
CPG(Reattached)	→			
CPG(Conference disconnected)	→			
		Conversation		
REL	→		→	BYE
RLC	←		←	200 OK BYE

TP604004	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.14** **ITU-T Rec Q.734.1 [34], clause 1.7**
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 5/10	
Test purpose:	*No mapping of generic notifications no change the session state* Ensure that the MGCF can receive in a CPG the Generic notifications is "other party added" or "other party isolated" or "other party reattached" or "other party split" or " conference floating" or " other party disconnected" and there is no mapping on the SIP side and the call is not disrupted. No mapping, no disrupting the SIP procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	CPG: Generic notification = Conference established CPG: Generic notification = isolated CPG: Generic notification = reattached CPG: Generic notification = Conference disconnected	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation CPG(Conference established) → CPG(other party added) → CPG(other party isolated) → CPG(other party reattached) → CPG(other party split) → CPG(other party disconnected) → CPG(Conference floating) → CPG(Conference disconnected) → Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP604005	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.1.1.1**
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "conference established"* Upon the receipt of a conference information document with the <conference-state-type> element *active* is set to "true", the MGCF shall send a CPG message to the CS side with a notification "***conference established***".	
SIP Parameter values:	NOTIFY 1: Event contains **conference**; Subscription-State contains **active; expires=xxxx** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>2</user-count> if present <active>**true**</active> if present <*users*> <*user* entity=ISUPx URI state="full" <endpoint entity=endpoint ISUPx URI <status>connected</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status> <*user* entity=SIPx URI state="full" <endpoint entity=endpoint SIPx URI <status>connected</status> <joining-method>dialed-in</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:	CPG(Conference established)	
Comments:	**ISUP** **MGCF** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ACM ← ← 200 OK INVITE CPG(Conference established) ← ← NOTIFY 1 → 200 OK NOTIFY REL → → BYE RLC ← ← 200 OK BYE	

TP604006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.1.1.1**
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "conference established"* Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to "connected" and it was not set to "on-hold" before and the Contact URI in the element entity is not the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "other party added".	
SIP Parameter values:	NOTIFY 1: see test case TP604006 NOTIFY 2: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>y</user-count> if present <*users*> <*user* entity=SIPx URI state="full" <endpoint entity=endpoint SIPx URI <status>**connected**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:	CPG(other party added)	
Comments:	**ISUP** **MGCF** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ACM ← ← 200 OK INVITE CPG(Conference established) ← ← NOTIFY 1 → 200 OK NOTIFY CPG(other party added) ← ← NOTIFY 2 → 200 OK NOTIFY REL → → BYE RLC ← ← 200 OK BYE	

TP604007	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.1.1.1**
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "isolated"* Upon the receipt of a conference information document with the <endpoint-type> and the element status of endpoint-status-type is set to "on-hold" and it was set to "connected" before and the Contact URI in the element entity is the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "**isolated**".	
SIP Parameter values:	NOTIFY 1: see test case TP604006 NOTIFY 2: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>2</user-count> if present <*users*> <*user* entity=ISUPx URI state="full" <endpoint entity=endpoint ISUPx URI <status>**on-hold**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:	CPG(isolated)	
Comments:	**ISUP** **MGCF** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ACM ← ← 200 OK INVITE CPG(Conference established) ← ← NOTIFY 1 → 200 OK NOTIFY CPG(isolated) ← ← NOTIFY 2 → 200 OK NOTIFY REL → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><td>TP604008</td><td>SIP reference: RFC 3261 [6]</td><td>ISUP reference:
ES 283 027 [1], clause 7.4.1.1.1</td></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/SS/CONF/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 1/1</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Conference notification information is mapped into "other party isolated"</i>
Upon the receipt of a conference information document with the <endpoint-type> and the element <i>status of endpoint-status-type</i> is set to "on-hold" and it was set to "connected" before and the Contact URI in the element <i>entity</i> is not the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "<i>other party isolated</i>".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">NOTIFY 1: see test case TP604006
NOTIFY 2: Event contains conference; Subscription-State contains active
application/conference-info+xml:
<<i>conference-info</i>>
entity=conference URI state="full" version="x"
<<i>conference-state</i>>
<user-count>2</user-count> if present
<<i>users</i>>
<<i>user</i> entity=SIPx URI state="full"
<endpoint entity=endpoint SIPx URI
<status>on-hold</status>
<joining-method>dialed-out</ joining-method>
<media id="1"
<status>sendrecv</status></td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">CPG(other party isolated)</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><td>ISUP</td><td></td><td>MGCF</td><td></td><td>SIP</td></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td>CPG(Conference established)</td><td>←</td><td></td><td>←</td><td>NOTIFY 1</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK NOTIFY</td></tr>
<tr><td>CPG(other party isolated)</td><td>←</td><td></td><td>←</td><td>NOTIFY 2</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK NOTIFY</td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>
</td></tr>
</table>

TP604009	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.1.1.1**
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "reattached"* Upon the receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* is set to "connected" and it was set to "on-hold" before and the Contact URI in the element *entity* is the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "***reattached***".	
SIP Parameter values:	NOTIFY 1: see test case TP604006 NOTIFY 2: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>2</user-count> if present <*users*> <*user* entity=ISUPx URI state="full" <endpoint entity=endpoint ISUPx URI <status>**on-hold**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status> NOTIFY 3: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>2</user-count> if present <*users*> <*user* entity=ISUPx URI state="full" <endpoint entity=endpoint ISUPx URI <status>**connected**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:	CPG(reattached)	
Comments:	**ISUP** **MGCF** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ACM ← ← 200 OK INVITE CPG(Conference established) ← ← NOTIFY 1 → 200 OK NOTIFY CPG(isolated) ← ← NOTIFY 2 → 200 OK NOTIFY CPG(reattached) ← ← NOTIFY 3 → 200 OK NOTIFY REL → → BYE RLC ← ← 200 OK BYE	

TP604010	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.1.1.1**
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "reattached"* Upon the receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* is set to "connected" and it was set to "on-hold" before and the Contact URI in the element *entity* is not the address of the served PSTN/ISDN participant, the MGCF shall send a CPG message to the CS side with a notification "***other party reattached***".	
SIP Parameter values:	NOTIFY 1: see test case TP604006 NOTIFY 2: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>2</user-count> if present <*users*> <*user* entity=SIPx URI state="full" <endpoint entity=endpoint SIPx URI <status>**on-hold**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status> NOTIFY 3: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>2</user-count> if present <*users*> <*user* entity=SIPx URI state="full" <endpoint entity=endpoint SIPx URI <status>**connected**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:	CPG(other party reattached)	
Comments:	**ISUP** **MGCF** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ACM ← ← 200 OK INVITE CPG(Conference established) ← ← NOTIFY 1 → 200 OK NOTIFY CPG(other party isolated) ← ← NOTIFY 2 → 200 OK NOTIFY CPG(other party reattached) ← ← NOTIFY 3 → 200 OK NOTIFY REL → → BYE RLC ← ← 200 OK BYE	

TP604011	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.1.1.1**
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is mapped into "other party disconnected"* Upon the receipt of a conference information document with the <endpoint-type> and the element *status of endpoint-status-type* is set to "disconnected" and the element *joining-method of joining-type* is not set to "focus-owner, the MGCF shall send a CPG message to the CS side with a notification "***other party disconnected***".	

SIP Parameter values:

```
NOTIFY 1: see test case TP604006
NOTIFY 2:    Event contains conference; Subscription-State contains active
             application/conference-info+xml:
                <conference-info>
                   entity=conference URI state="full" version="x"
                   <conference-state>
                      <user-count>y</user-count> if present
                   <users>
                      <user entity=SIPx URI state="full"
                         <endpoint entity=endpoint SIPx URI
                            <status>connected</status>
                            <joining-method>dialed-out</ joining-method>
                            <media id="1"
                               <status>sendrecv</status>
NOTIFY 3:    Event contains conference; Subscription-State contains active
             application/conference-info+xml:
                <conference-info>
                   entity=conference URI state="full" version="x"
                   <conference-state>
                      <user-count>1</user-count> if present
                   <users>
                      <user entity=SIPx URI state="full"
                         <endpoint entity=endpoint SIPx URI
                            <status>disconnected</status>
                            <joining-method>dialed-out</ joining-method>
                            <disconnection-method>booted<disconnection-method/>
                            <media id="1"
                               <status>sendrecv</status>
```

ISUP Parameter values:	CPG(other party disconnected)

Comments:

ISUP		**MGCF**		**SIP**
IAM	→		→	INVITE
ACM	←		←	180 Ringing
ACM	←		←	200 OK INVITE
CPG(Conference established)	←		←	NOTIFY 1
			→	200 OK NOTIFY
CPG(other party added)	←		←	NOTIFY 2
			→	200 OK NOTIFY
CPG(other party disconnected)	←		←	NOTIFY 3
			→	200 OK NOTIFY
REL	→		→	BYE
RLC	←		←	200 OK BYE

TP604012	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.1.1.1**
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:	NOT PICS 1/1	
ISUP selection criteria:		
Test purpose:	*Conference notification information is not mapped to PSTN* Upon the receipt of a conference information document the conference notification information is not mapped to the PSTN side. No NOTIFY is sent to the ISDN user.	
SIP Parameter values:	NOTIFY 1: Event contains **conference**; Subscription-State contains **active; expires=xxxx** NOTIFY 1: see test case TP604006 NOTIFY 2: Event contains **conference**; Subscription-State contains **active** application/conference-info+xml: <*conference-info*> entity=conference URI state="full" version="x" <*conference-state*> <user-count>y</user-count> if present <*users*> <*user* entity=SIPx URI state="full" <endpoint entity=endpoint SIPx URI <status>**connected**</status> <joining-method>dialed-out</ joining-method> <media id="1" <status>sendrecv</status>	
ISUP Parameter values:	CPG(other party added)	
Comments:	**ISUP** **MGCF** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ACM ← ← 200 OK INVITE ← NOTIFY 1 → 200 OK NOTIFY ← NOTIFY 2 → 200 OK NOTIFY REL → → BYE RLC ← ← 200 OK BYE	

TP604013	**SIP reference: RFC 3261 [6]**	**ISUP reference:**
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*The referring of MGCF is not possible when a call is established* Ensure that a REFER request received by the MGCF is not successful. The request is rjected with . 403 Forbidden. The CS -site is not affected.	
SIP Parameter values:	REFER: Request URI contained the **conference URI** Refer-To contains the URI of **ISUPx**, method=invite Referred-By contains SIP or tel URI of **SIPx**	
ISUP Parameter values:	CPG(Conference established)	
Comments:	**ISUP** **MGCF** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ACM ← ← 200 OK INVITE ← REFER → 403 Forbidden	

6.3.2.5 Three Party service (3PTY)

TP605001	**SIP reference: RFC 3261 [6]**	**NGN reference:** **ES 283 027 [1], clause 7.4.15**
TSS reference:	ISUP-SIP/SS/3PTY/	
SIP selection criteria:	PICS 8/2	
ISUP selection criteria:	PICS 5/5 AND PICS 5/18	
Test purpose:	*The media stream is resumed if a 3PTY is established* • Ensure that the SUT resumes the media stream put on hold while the GPG (hold) was recived and sends a re-INVITE containing an a-line in the SDP is set to "sendrecv" if a CPG (Conference established) was received	
SIP Parameter values:	SDP: a= **a_LINE_VA** (see table 45) or a line is omitted	
ISUP Parameter values:	CPG: Generic notification = remote hold CPG: Generic notification = **sendrecv**	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation CPG(hold) → → INVITE(sendonly) ← 200 OK INVITE(recvonly) → ACK CPG(Conference established) → → INVITE(sendrecv) ← 200 OK INVITE(sendrecv) → ACK CPG(Conference disconnected) → Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP605002	**SIP reference: RFC 3261 [6]**	**NGN reference:** **ES 283 027 [1], clause 7.4.15**
TSS reference:	ISUP-SIP/SS/3PTY /	
SIP selection criteria:	PICS 8/1	
ISUP selection criteria:	PICS 5/5 AND PICS 5/18	
Test purpose:	*Establish and disconnect a 3PTY session. SDP conveyed in an UPDATE request* • Ensure that the SUT resumes the media stream put on hold while the GPG (hold) was recived and sends an UPDATE containing an a-line in the SDP is set to "sendrecv" if a CPG (Conference established) was received	
SIP Parameter values:	SDP: a= **a_LINE_VA** (see table 45) or a line is omitted	
ISUP Parameter values:	CPG: Generic notification = remote hold CPG: Generic notification = **GEN_NOT_VALUE**	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation CPG(hold) → → UPDATE(sendonly) ← 200 OK UPDATE(recvonly) CPG(Conference established) → → UPDATE(sendrecv) ← 200 OK UPDATE(sendrecv) CPG(Conference disconnected) → Conversation REL → → BYE RLC ← ← 200 OK BYE	

Table 45

Values for test purpose TP605001, TP605002		
CPG→ **Generic notification** **GEN_NOT_VALUE**		**INVITE/UPDATE→** **SDP attribute line** **a_LINE_VA**
VA_01	Conference established	sendrecv, or recvonly
VA_02	Conference disconnected	sendrecv or recvonly

TP605003	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.13 ITU-T Rec Q.734.2 [35], clause 2.7**
TSS reference:	ISUP-SIP/SS/3PTY/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 5/18	
Test purpose:	*Interworking of "conference established" and "Conference disconnected" not supported* Ensure that the SUT on receipt of a CPG message due to the 3PTY supplementary service, the Generic notification indicator with the value. No mapping, no disrupting the SIP procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	CPG: Generic notification = remote hold CPG: Generic notification = Conference established CPG: Generic notification = Conference disconnected	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation CPG(hold) → CPG(Conference established) → CPG(Conference disconnected) → Conversation REL → → BYE RLC ← ← 200 OK BYE	

6.3.2.6 Connected line identification (COL)

TP606001	**SIP reference: RFC 3261 [6]**	**ES 283 027 [1], clause 7.4.2**
TSS reference:	ISUP-SIP/SS/COL /	
SIP selection criteria:	NOT PICS 5/3	
ISUP selection criteria:		
Test purpose:	*IAM with OFCI "connected line request" received, no mapping* Ensure that the SUT if the IAM is received with an optional forward call indicator, connected line requested, continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	IAM: Optional Forward call indicator "Connected line request"	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Conversation** REL → → BYE RLC ← ← 200 OK BYE	

TP606002	**SIP reference: RFC 3261 [6]**			**ES 283 027 [1], clause 7.4.2**	
TSS reference:	ISUP-SIP/SS/COL /				
SIP selection criteria:	PICS 5/3				
ISUP selection criteria:					
Test purpose:	*IAM with oFCi "connected line request" received, INVITE is sent contains the "from-change" tag in the Supported header* Ensure that the SUT if the IAM is received with an optional forward call indicator, connected line requested, the "from-change" tag is included in the Supported header in the sent INVITE.				
SIP Parameter values:	INVITE: Supported: from-change				
ISUP Parameter values:	IAM: Optional Forward call indicator "Connected line request"				
	ISUP		**SUT**		**SIP**
	IAM	→		→	INVITE
	ACM	←		←	180 Ringing
	ANM	←		←	200 OK INVITE
				→	ACK
			Conversation		
	REL	→		→	BYE
	RLC	←		←	200 OK BYE

TP606003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.2**
TSS reference:	ISUP-SIP/SS/COL /	
SIP selection criteria:	PICS 5/3	
ISUP selection criteria:	NOT PICS 1/5	
Test purpose:	*The P-Asserted-Identity is mapped into the connected number "national (significant) number"* Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the „Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with the P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and the no Privacy header field was received or a Privacy header field was received and the priv-value is set to "none" in the ANM or CON is included the **Connected number Parameter**. If CC encoded in the URI is **equal** to the CC of the country where MGCF is located AND the next BICC/ISUP node is located in the same country then Address presentation restricted parameter = **presentation allowed** Nature of address indicator = **National (significant) number** Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network Provided Address signals in the format: **NDC+SN** Generic number parameter not present	
SIP Parameter values:	1XX or 2XX response: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN	
ISUP Parameter values:	IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" = "requested" ANM; Connected number parameter Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT	
	ISUP **MGCF** **SIP** IAM → → INVITE ← SIP_MESSAGE_VA **CASE A** ACM ← ANM ← **CASE B** CON ← **Conversation** REL → → BYE RLC ← ← 200 OK BYE	

TP606004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.2**
TSS reference:	ISUP-SIP/SS/COL /	
SIP selection criteria:	PICS 5/3	
ISUP selection criteria:	PICS 1/5	
Test purpose:	*The P-Asserted-Identity is mapped into the connected number "international number"* Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the „Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with the P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and the no Privacy header field was received or a Privacy header field was received and the priv-value is set to "none" in the ANM or CON is included the **Connected number Parameter**. If CC encoded in the URI is **not equal** to the CC of the country where MGCF is located AND the next BICC/ISUP node is located in the same country then Address presentation restricted parameter = **Presentation allowed** Nature of address indicator = **International number** Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network Provided Address signals in the format: **CC+NDC+SN** Generic number parameter not present	
SIP Parameter values:	1XX or 2XX response: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN	
ISUP Parameter values:	IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" = "requested" ANM; Connected number parameter Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT	
	ISUP **MGCF** **SIP** IAM → → INVITE ← SIP_MESSAGE_VA **CASE A** ACM ← ANM ← **CASE B** CON ← **Conversation** REL → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP606005</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.4.2</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/SS/COL /</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2">PICS 5/3</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 1/5</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>P-Asserted-Identity not received, a connected number "address not available" is sent</i>
Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the „Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with
no P-Asserted-Identity header field
In the ANM or CON is included the Connected number Parameter.
Address presentation restricted parameter = Address not available
Screening indicator = Network Provided
Address signals omitted
Generic number parameter not present</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">1XX or 2XX response: P-Asserted-Identity header field is not present</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" = "requested"
ANM or CON
Connected number parameter
Address presentation restricted parameter = '10'B
Nature of address indicator = '0000000'B
Numbering plan indicator = '000'B
Screening indicator = '11'B
Address signals = not presented</td></tr>
<tr><td></td><td colspan="2">ISUP MGCF SIP
IAM → → INVITE
← SIP_MESSAGE_VA
CASE A
ACM ←
ANM ←
CASE B
CON ←
Conversation
REL → → BYE
RLC ← ← 200 OK BYE</td></tr>
</table>

Values for tests purposes TP101003 to TP101005	
VA_01	180 Ringing
VA_02	183 Session progress
VA_03	200 OK

TP606006	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.2**
TSS reference:	ISUP-SIP/SS/COL /	
SIP selection criteria:	PICS 5/3	
ISUP selection criteria:	NOT PICS 1/5	
Test purpose:	*Interworking of P-Asserted-Identity. Convert the user portion into the national format* Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the „Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with the P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and the a Privacy header field was received and the priv-value is set to PRIV_VALUE in the ANM or CON is included the **Connected number Parameter**. If CC encoded in the URI is **equal** to the CC of the country where MGCF is located AND the next BICC/ISUP node is located in the same country then Address presentation restricted parameter = **Presentation restricted** Nature of address indicator = **National (significant) number** Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network Provided Address signals in the format: **NDC+SN** Generic number parameter not present	
SIP Parameter values:	1XX or 2XX response: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN	
ISUP Parameter values:	IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" = "requested" ANM or CON Connected number parameter Address presentation restricted parameter = '01'B Nature of address indicator = "0000011"B Numbering plan indicator = "001"B Screening indicator = "11"B Address signals = PIXIT	

ISUP		**MGCF**		**SIP**
IAM	→		→	INVITE
			←	SIP_MESSAGE_VA
CASE A				
ACM	←			
ANM	←			
CASE B				
CON	←			
		Conversation		
REL	→		→	BYE
RLC	←		←	200 OK BYE

TP606007	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.2**
TSS reference:	ISUP-SIP/SS/COL /	
SIP selection criteria:	PICS 5/3	
ISUP selection criteria:	PICS 1/5	
Test purpose:	*Interworking of P-Asserted-Identity. Convert the user portion into the international format* Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the „Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with the P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received and the a Privacy header field was received and the priv-value is set to PRIV_VALUE in the ANM or CON is included the **Connected number Parameter**. If CC encoded in the URI is **not equal** to the CC of the country where MGCF is located AND the next BICC/ISUP node is located in the same country then Address presentation restricted parameter = **Presentation restricted** Nature of address indicator = **International number** Numbering plan indicator = ISDN/Telephony numbering plan Screening indicator = Network Provided Address signals in the format: **CC+NDC+SN** **Generic number parameter** not present	
SIP Parameter values:	1XX or 2XX response: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN	
ISUP Parameter values:	IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" = "requested" ANM or CON **Connected number parameter** Address presentation restricted parameter = '01'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT	
	ISUP **MGCF** **SIP** IAM → → INVITE ← SIP_MESSAGE_VA **CASE A** ACM ← ANM ← **CASE B** CON ← **Conversation** REL → → BYE RLC ← ← 200 OK BYE	

TP606008	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clauses 7.4.2 and 7.5.2**
TSS reference:	ISUP-SIP/SS/COL /	
SIP selection criteria:	PICS 5/3	
ISUP selection criteria:		
Test purpose:	*Interworking of From header in the UPDATE. An additional connected number is sent in the ANM or CON* Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the „Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with the option tag "from-change" is received a Privacy header field was received and the priv-value is set to PRIV_VALUE An UPDATE request is received containing a changed From header field containing a URI with an identity in the format "+" CC+ NDC+ SN then - map the From header field received in the UPDATE request to the Generic number in the ANM or CON In the ANM or CON is included the **Generic number parameter** Number Qualifier = additional connected number Address presentation restricted parameter = **Presentation allowed** Numbering plan indicator = ISDN numbering plan Screening indicator = user provided, not verified Address signals = derived from the From header in the UPDATE	
SIP Parameter values:	INVITE: Supported: from-change 1XX or 2XX response: P-Asserted-Identity header field URI in the format "+"CC+NDC+SN, Supported: from-change UPDATE: From header in the format "+"CC+NDC+SN	
ISUP Parameter values:	IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" = "requested" ANM or CON **Genericnumber** Number Qualifier = "00000101"B Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = derived from the From header in the UPDATE	

	ISUP		**MGCF**		**SIP**
	IAM	→		→	INVITE
				←	SIP_MESSAGE_VA
	CASE A				
	ACM	←			
				←	UPDATE
	ANM	←		→	200 OK UPDATE
	CASE B				
				←	UPDATE
	CON	←		→	200 OK UPDATE
			Conversation		
	REL	→		→	BYE
	RLC	←		←	200 OK BYE

TP606009	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clauses 7.4.2 and 7.5.2**
TSS reference:	ISUP-SIP/SS/COL /	
SIP selection criteria:	PICS 5/22	

<table>
<tr><th>TP606009</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
ES 283 027 [1], clauses 7.4.2 and 7.5.2</th></tr>
<tr><td>**ISUP selection criteria:**</td><td colspan="5"></td></tr>
<tr><td>**Test purpose:**</td><td colspan="5">*Interworking of P-Asserted-Identity and From header. The Connected number and the additional connected number is presentation allowed*

Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the „Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested",
on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with

the P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN and the option tag "from-change" is received
a Privacy header field is not included

An UPDATE request is received containing a changed From header field containing a URI with an identity in the format "+" CC+ NDC+ SN then
- map the From header field received in the UPDATE request to the Generic number in the ANM

In the ANM or CON is included the **Connected number Parameter**.
Numbering plan indicator = ISDN/Telephony numbering plan
Address presentation restricted parameter = Presentation allowed
Screening indicator = Network Provided
Address signals derived from the P-Asserted-Identity in the 200 OK INVITE

In the ANM or CON is included the **Generic number parameter**
Number Qualifier = additional connected number
Address presentation restricted parameter = Presentation allowed
Numbering plan indicator = ISDN numbering plan
Screening indicator = user provided, not verified
Address signals derived from the From header in the UPDATE</td></tr>
<tr><td>**SIP Parameter values:**</td><td colspan="5">INVITE: Supported: from-change
1XX or 2XX response: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN
UPDATE: From header, P-Asserted-Identity</td></tr>
<tr><td>**ISUP Parameter values:**</td><td colspan="5">IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" = "requested"

ANM or CON
Connected number parameter
Address presentation restricted parameter = "00"B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT
Generic number parameter
Number Qualifier = "00000101"B
Address presentation restricted parameter = "00"B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = derived from the From header in the UPDATE</td></tr>
<tr><td></td><td>**ISUP**</td><td></td><td>**MGCF**</td><td></td><td>**SIP**</td></tr>
<tr><td></td><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td></td><td>**CASE A**</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>ACM</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td>←</td><td>UPDATE</td></tr>
<tr><td></td><td>ANM</td><td>←</td><td></td><td>→</td><td>200 OK UPDATE</td></tr>
<tr><td></td><td>**CASE B**</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td>←</td><td>UPDATE</td></tr>
<tr><td></td><td>CON</td><td>←</td><td></td><td>→</td><td>200 OK UPDATE</td></tr>
<tr><td></td><td></td><td></td><td>**Conversation**</td><td></td><td></td></tr>
<tr><td></td><td>REL</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td></td><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>

<table>
<tr><th>TP606010</th><th colspan="3">SIP reference: RFC 3261 [6]</th><th colspan="2">ISUP reference:
ES 283 027 [1], clauses 7.4.2 and 7.5.2</th></tr>
<tr><td>TSS reference:</td><td colspan="5">ISUP-SIP/SS/COL /</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="5">PICS 5/22</td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="5"></td></tr>
<tr><td>Test purpose:</td><td colspan="5"><i>Interworking of P-Asserted-Identity to the connected number if no UPDATE was received</i>

Ensure that the SUT in Idle state, on receipt of an IAM message where the COLP service has been requested by the calling party by parsing the „Optional Forward Call Indicators" field and the "Connected Line Identity Request indicator" is set to "requested",
on receipt of a 1XX or 2XX message defined as SIP_MESSAGE_VA with

the P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN and the option tag "from-change" is received
a Privacy header field is not included

When the 200 OK was received, start timer T_{TIR1}

An UPDATE request is not received

After T_{TIR1} was expired the ANM or CON is sent

In the ANM or CON is included the Connected number Parameter.
Numbering plan indicator = ISDN/Telephony numbering plan
Address presentation restricted parameter = Presentation allowed
Screening indicator = Network Provided
Address signals derived from the P-Asserted-Identity in the 200 OK INVITE</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="5">INVITE: Supported: from-change
1XX or 2XX response: P-Asserted-Identity header field Tel URL containing an URI in the format "+"CC+NDC+SN</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="5">IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" = "requested"

ANM or CON
Connected number parameter
Address presentation restricted parameter = "00"B
Numbering plan indicator = '001'B
Screening indicator = '11'B
Address signals = PIXIT</td></tr>
<tr><td></td><td>ISUP</td><td></td><td>MGCF</td><td></td><td>SIP</td></tr>
<tr><td></td><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td></td><td>←</td><td>SIP_MESSAGE_VA</td></tr>
<tr><td></td><td>CASE A</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td>ACM</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>T_{TIR1}</td><td></td><td></td></tr>
<tr><td></td><td>ANM</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td>CASE B</td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>T_{TIR1}</td><td></td><td></td></tr>
<tr><td></td><td>CON</td><td>←</td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td></td><td>REL</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td></td><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>

Values for tests purposes TP102005 to TP102010	
VA_01	180 Ringing
VA_02	183 Session progress
VA_03	200 OK

Values for test purpose TP102006 to TP102009	
VA	**PRIV_VALUE**
VA_1	Id
VA_2	User
VA_3	Header

6.3.2.7 Sub-addressing (SUB)

TP607001	**CW Reference:** **ES 283 027 [1], clause 4.7.4.5.2**	**Selection criteria:** **PICS 5/8**
TSS reference:	ISUP-SIP/SS/SUB/	
Preconditions:		
Test purpose:	*The calling party subaddress is mapped in the isub parameter of the P-Asserted-Identity* Ensure that the calling party subaddress in the ATP parameter of the received IAM is interworked in the isub parameter of the P-Asserted-Identity in the sent INVITE, if the Type of Subbaddress is set to "0 0 0" "NSAP.	
SIP Parameter values:	INVITE: P-Asserted-Identity: sip: user part; isub=<subaddress>@hostportion	
ISUP Parameter values:	IAM: ATP(Calling party subaddress)	
Comments:	ISUP MGCF SIP IAM(ATP) → → INVITE ← 100 Trying ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK Communication REL → → BYE RLC ← ← 200 OK BYE	

TP607002	**CW Reference:** **4.7.4.5.2**	**Selection criteria:** **PICS 5/8**
TSS reference:	ISUP-SIP/SS/SUB/	
Preconditions:		
Test purpose:	*The called party subaddress is mapped in the isub parameter of the Request URI* Ensure that the called party subaddress in the ATP parameter of the received IAM is interworked in the isub parameter of the Request URI in the sent INVITE, if the Type of Subbaddress is set to "0 0 0" "NSAP.	
SIP Parameter values:	INVITE: Request URI: sip: user part; isub=<subaddress>@hostportion	
ISUP Parameter values:	IAM: ATP(Called party subaddress)	
Comments:	**ISUP** **MGCF** **SIP** IAM(ATP) → → INVITE ← 100 Trying ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK Communication REL → → BYE RLC ← ← 200 OK BYE	

TP607003	**CW Reference:** **4.7.4.5.2**	**Selection criteria:** **PICS 5/8**
TSS reference:	ISUP-SIP/SS/SUB/	
Preconditions:		
Test purpose:	*The isub parameter of the P-Asserted-Identity in the 200 OK INVITE is mapped in the connected subaddress in the ANM* Ensure that the isub parameter in the P-Asserted-Identity of the received 200 OK INVITE is interworked in the Connected subaddress contained in an ATP parameter in the sent ANMOBCI. The Type of Subbaddress is set set to "0 0 0" "NSAP (ITU-T Recommendation X.213 [29] and ISO/IEC 8348 [30] Add.2)"	
SIP Parameter values:	INVITE: supported: from-change 200 OK INVITE: P-Asserted-IDENTITY: sip: user part; isub=<subaddress>@hostportion	
ISUP Parameter values:	IAM: oFCi: connected line request ANM: ATP(Connected subaddress)	
Comments:	**ISUP** **MGCF** **SIP** IAM → → INVITE ← 100 Trying ACM ← ← 180 Ringing ← 200 OK INVITE → ACK ANM(ATP) ← ← UPDATE → 200 OK UPDATE Communication REL → → BYE RLC ← ← 200 OK BYE	

TP607004	**CW Reference:** **4.7.4.5.2**	**Selection criteria:** **NOT PICS 5/8**
TSS reference:	ISUP-SIP/SS/SUB/	
Preconditions:		
Test purpose:	*The calling party subaddress is not mapped in the isub parameter of the P-Asserted-Identity* Ensure that the calling party subaddress in the ATP parameter of the received IAM is not interworked in the isub parameter of the From header in the sent INVITE, if the Type of Subbaddress is not set to "0 0 0" "NSAP.	
SIP Parameter values:	INVITE: P-Asserted-Identity: sip: user part; isub=<subaddress>@hostportion	
ISUP Parameter values:	IAM: ATP(no Calling party subaddress)	
Comments:	**ISUP** **MGCF** **SIP** IAM(ATP) → → INVITE ← 100 Trying ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK Communication REL → → BYE RLC ← ← 200 OK BYE	

TP607005	**CW Reference:** **4.7.4.5.2**	**Selection criteria:** **NOT PICS 5/8**
TSS reference:	ISUP-SIP/SS/SUB/	
Preconditions:		
Test purpose:	*The called party subaddress is not mapped in the isub parameter of the Request URI* Ensure that the called party subaddress in the ATP parameter of the received IAM is not interworked in the isub parameter of the Request URI in the sent INVITE, if the Type of Subbaddress is not set to "0 0 0" "NSAP.	
SIP Parameter values:	INVITE: Request URI: sip: user part; isub=<subaddress>@hostportion	
ISUP Parameter values:	IAM: ATP(no Called party subaddress)	
Comments:	**ISUP** **MGCF** **SIP** IAM(ATP) → → INVITE ← 100 Trying ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK Communication REL → → BYE RLC ← ← 200 OK BYE	

6.3.2.8 Closed user group (CUG)

TP608001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.16**
TSS reference:	ISUP-SIP/SS/CUG/	
SIP selection criteria:	NOT PICS 5/7	
ISUP selection criteria:		
Test purpose:	*SIP network dows not support CUG, CUG with outgoing access allowed is interworked in a normal call.* Ensure that the SUT if an IAM is received with Optional forward call indicator, CUG call indicator coded as "**CUG call with outgoing access**" and CUG interlock code or CUG call indicator coded as "Non CUG call" or Optional forward call indicator is absent, the SIP signalling procedure is not disrupted.	
SIP Parameter values:	No mapping	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP608002	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.16**
TSS reference:	ISUP-SIP/SS/CUG/	
SIP selection criteria:	NOT PICS 5/7	
ISUP selection criteria:		
Test purpose:	*SIP network dows not support CUG, CUG with outgoing access not allowed is rejected.* Ensure that the SUT if an IAM is received with Optional forward call indicator, CUG call indicator coded as "**CUG call without outgoing access**" and CUG interlock code, a **REL** is sent. No INVITE is sent into the SIP network.	
SIP Parameter values:	No action	
ISUP Parameter values:	REL: Cause #29	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → REL ← RLC →	

TP608003	**SIP reference: RFC 3261 [6]**	**ISUP reference: 7.4.1.2**
TSS reference:	ISUP-SIP/SS/CUG/	
SIP selection criteria:	PICS 5/7	
ISUP selection criteria:		
Test purpose:	*SIP network supports CUG. CUG call indicator value "10" received.* Ensure that Optional Forward Call Indicator Parameter **CUG call indicator** is mapped into <cug> < cugCommunicationIndicator>, the Closed user group interlock code Parameter **Network indentity** is mapped into <cug> <networkIndicator> and the **Binary code** is mapped into the <cug> <cugInterlockBinaryCode>.	
SIP Parameter values:	INVITE: <cug> <networkIndicator>[derived from IAM **Network indentity**]</networkIndicator> <cugInterlockBinaryCode>[derived from IAM **Binary code**]</cugInterlockBinaryCode> <cugCommunicationIndicator>10</cugCommunicationIndicator> </cug>	
ISUP Parameter values:	IAM: Optional Forward Call Indicator CUG call indicator = "10" Closed User Group interlock code **Binary code** derived from INVITE XML body <**cugInterlockBinaryCode**> **Network indentity** derived from INVITE XML body <**networkIndicator**>	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP608004	**SIP reference: RFC 3261 [6]**	**ISUP reference: 7.4.1.2**
TSS reference:	ISUP-SIP/SS/CUG/	
SIP selection criteria:	PICS 5/7	
ISUP selection criteria:		
Test purpose:	*SIP network supports CUG. CUG call indicator value "11" received.* Ensure that Optional Forward Call Indicator Parameter **CUG call indicator** is mapped into <cug> < cugCommunicationIndicator>, the Closed user group interlock code Parameter **Network indentity** is mapped into <cug> <networkIndicator> and the **Binary code** is mapped into the <cug> <cugInterlockBinaryCode>.	
SIP Parameter values:	INVITE: <cug> <networkIndicator>[derived from IAM **Network indentity**]</networkIndicator> <cugInterlockBinaryCode>[derived from IAM **Binary code**]</cugInterlockBinaryCode> <cugCommunicationIndicator>11</cugCommunicationIndicator> </cug>	
ISUP Parameter values:	IAM: Optional Forward Call Indicator CUG call indicator = "11" Closed User Group interlock code **Binary code** derived from INVITE XML body <**cugInterlockBinaryCode**> **Network indentity** derived from INVITE XML body <**networkIndicator**>	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

6.3.2.9 Call diversion (CDIV)

TP609001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.6**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	NOT PICS 5/12 AND NOT PICS 5/13 AND NOT PICS 5/14 AND NOT PICS 5/15	
ISUP selection criteria:		
Test purpose:	*CDIV parameter not mapped* Ensure that the SUT if the IAM is received with **Redirecting number, original called number and redirection information**, continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	IAM: Redirecting number, Original called number, Redirection information	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP609002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*IAM with Original call number and redirecting number Presentation allowed received* Ensure that the SUT if the IAM is received with **Redirecting number, original called number** Presentation allowed and **redirection information** Presentation allowed, the redirection counter value is "1", an INVITE is sent containing a History-Info header. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.	
SIP Parameter values:	INVITE: History-Info header hi-targeted-to-uri Redirecting number; index=1, *or* hi-targeted-to-uri Redirecting number?Privacy=none/absent; index=1, hi-targeted-to uri diverted to user; cause=**Cause_value**; index=1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion" Redirection counter = 1 Redirecting indicator = 3 Redirecting reason = ISUP_RR Original called number Presentation restriction: Presentation allowed Redirecting number Presentation restriction: Presentation allowed	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC 200 OK BYE ← ←	

IAM		INVITE	
ISUP Parameter	**Source value of parameter field**	**SIP component**	**Derived value of header field**
Redirection Information	**Redirecting reason**	**History-Info header**	**Cause parameter**
	unknown '0000'B	**Cause_value**	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*IAM with Original call number Presentation restricted and redirecting number Presentation allwed received* Ensure that the SUT if the IAM is received with **Redirecting number, original called number** Presentation restricted and **redirection information** Presentation allowed, the redirection counter value is "1", an INVITE is sent containing a History-Info header. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.	
SIP Parameter values:	INVITE: History-Info header hi-targeted-to-uri Redirecting number?Privacy=history; index=1, hi-targeted-to uri diverted to user; cause=**Cause_value**; index=1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion" Redirection counter = 1 Redirecting indicator = 3 Redirecting reason = ISUP_RR Original called number Presentation restriction: Presentation restricted Redirecting number Presentation restriction: Presentation allowed	
Comments:	**ISUP** **SUT** **SIP** IAM ➔ ➔ INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE ➔ ACK **Communication** REL ➔ ➔ BYE RLC 200 OK BYE ← ←	

IAM		**INVITE**	
ISUP Parameter or IE	**Source value of parameter field**	**SIP component**	**Derived value of header field**
Redirection Information	Redirecting reason	**History-Info header**	**Cause parameter**
	unknown '0000'B	**Cause_value**	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609004	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*IAM with Original call number Presentation allowed and redirecting number Presentation allowed restricted* Ensure that the SUT if the IAM is received with **Redirecting number, original called number** Presentation allowed and **redirection information** Presentation restricted, the redirection counter value is "1", an INVITE is sent containing a History-Info header. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.	
SIP Parameter values:	INVITE: History-Info header hi-targeted-to-uri Redirecting number?Privacy=history; index=1, hi-targeted-to uri diverted to user; cause=**Cause_value**; index=1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion" Redirection counter = 1 Redirecting indicator = 4 Redirecting reason = ISUP_RR Original called number Presentation restriction: Presentation allowed Redirecting number Presentation restriction: Presentation restricted	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC ← ← 200 OK BYE	

IAM		**INVITE**	
ISUP Parameter or IE	**Source value of parameter field**	**SIP component**	**Derived value of header field**
Redirection Information	**Redirecting reason**	**History-Info header**	**Cause parameter**
	unknown '0000'B	**Cause_value**	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609005	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*IAM with Original call number Presentation allowed and redirecting number Presentation allowed restricted* Ensure that the SUT if the IAM is received with **Redirecting number, original called number** Presentation restricted and **redirection information** Presentation restricted, the redirection counter value is "1", an INVITE is sent containing a History-Info header. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.	
SIP Parameter values:	INVITE: History-Info header hi-targeted-to-uri Redirecting number?Privacy=history; index=1, hi-targeted-to uri diverted to user; cause=**Cause_value**; index=1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion" Redirection counter = 1 Redirecting indicator = 4 Redirecting reason = ISUP_RR Original called number Presentation restriction: Presentation restricted Redirecting number Presentation restriction: Presentation restricted	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC 200 OK BYE ← ←	

IAM		**INVITE**	
ISUP Parameter or IE	**Source value of parameter field**	**SIP component**	**Derived value of header field**
Redirection Information	**Redirecting reason**	**History-Info header**	**Cause parameter**
	unknown '0000'B	**Cause_value**	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609006	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*IAM with Original call number and redirecting number Presentation allowed received, Redirecting indicator indicates "all redirection information presentation restricted"* Ensure that the SUT if the IAM is received with **Redirecting number, original called number** Presentation allowed and **redirection information** Presentation restricted, the redirection counter value is "1", an INVITE is sent containing a History-Info header. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.	
SIP Parameter values:	INVITE: History-Info header hi-targeted-to-uri Redirecting number; index=1, *or* hi-targeted-to-uri Redirecting number?Privacy=history; index=1, hi-targeted-to uri diverted to user; cause=**Cause_value**; index=1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion" Redirection counter = 1 Redirecting indicator = 4 Redirecting reason = ISUP_RR Original called number Presentation restriction: Presentation allowed Redirecting number Presentation restriction: Presentation allowed	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC 200 OK BYE ← ←	

IAM		**INVITE**	
ISUP Parameter	**Source value of parameter field**	**SIP component**	**Derived value of header field**
Redirection Information	**Redirecting reason**	**History-Info header**	**Cause parameter**
	unknown '0000'B	**Cause_value**	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609007	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*IAM with Original call number and redirecting number Presentation allowed received, Redirection counter value 2* Ensure that the SUT if the IAM is received with **Redirecting number, original called number** Presentation allowed and **redirection information** Presentation allowed, the redirection counter value is "2", an INVITE is sent containing a History-Info header. The Original called number is contained in the hi-tatgeted-to uri in the index 1. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1.1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.	
SIP Parameter values:	INVITE: History-Info header hi-targeted-to-uri Original called number; index=1, hi-targeted-to-uri Redirecting number; cause=**302**; index=1.1, hi-targeted-to-uri called party number; cause= **Cause_value**; index=1.1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion" Redirection counter = 2 Redirecting indicator = 3 Redirecting reason = ISUP_RR Original called number Presentation restriction: Presentation allowed Redirecting number Presentation restriction: Presentation allowed	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC 200 OK BYE ← ←	

IAM		**INVITE**	
ISUP Parameter or	**Source value of parameter field**	**SIP component**	**Derived value of header field**
Redirection Information ISUP_RR	**Redirecting reason**	**History-Info header**	**Cause parameter**
	unknown '0000'B	**Cause_value**	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609008	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*IAM with Original call number Presentation restricted and redirecting number Presentation allowed received, Redirection counter value 2* Ensure that the SUT if the IAM is received with **Redirecting number, original called number** Presentation restricted and **redirection information** Presentation allowed, the redirection counter value is "2", an INVITE is sent containing a History-Info header. The Original called number is contained in the hi-tatgeted-to uri in the index 1. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1.1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.	
SIP Parameter values:	INVITE: History-Info header hi-targeted-to-uri Original called number?Privacy=history; index=1, hi-targeted-to-uri Redirecting number; cause=**302**; index=1.1, hi-targeted-to-uri called party number; cause= **Cause_value**; index=1.1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion" Redirection counter = 2 Redirecting indicator = 3 Redirecting reason = ISUP_RR Original called number Presentation restriction: Presentation restricted Redirecting number Presentation restriction: Presentation allowed	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC ← ← 200 OK BYE	

IAM		**INVITE**	
ISUP Parameter or	**Source value of parameter field**	**SIP component**	**Derived value of header field**
Redirection Information ISUP_RR	**Redirecting reason**	**History-Info header**	**Cause parameter**
	unknown '0000'B	**Cause_value**	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609009	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*IAM with Original call number Presentation allowed and redirecting number Presentation restricted received, Redirection counter value 2* Ensure that the SUT if the IAM is received with **Redirecting number, original called number** Presentation allowed and **redirection information** Presentation restricted, the redirection counter value is "2", an INVITE is sent containing a History-Info header. The Original called number is contained in the hi-tatgeted-to uri in the index 1. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1.1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.	
SIP Parameter values:	INVITE: History-Info header hi-targeted-to-uri Original called number; index=1, hi-targeted-to-uri Redirecting number; ?Privacy=history;cause=**302**; index=1.1, hi-targeted-to-uri called party number; cause=**Cause_value**; index=1.1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion" Redirection counter = 2 Redirecting indicator = 4 Redirecting reason = ISUP_RR Original called number Presentation restriction: Presentation allowed Redirecting number Presentation restriction: Presentation restricted	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC 200 OK BYE ← ←	

IAM		**INVITE**	
ISUP Parameter or	**Source value of parameter field**	**SIP component**	**Derived value of header field**
Redirection Information ISUP_RR	**Redirecting reason**	**History-Info header**	**Cause parameter**
	unknown '0000'B	**Cause_value**	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609010	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*IAM with Original call number and redirecting number Presentation restricted received, Redirection counter value 2* Ensure that the SUT if the IAM is received with **Redirecting number, original called number** Presentation restricted and **redirection information** Presentation restricted, the redirection counter value is "2", an INVITE is sent containing a History-Info header. The Original called number is contained in the hi-tatgeted-to uri in the index 1. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1.1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.	
SIP Parameter values:	INVITE: History-Info header hi-targeted-to-uri Original called number?Privacy=history index=1, hi-targeted-to-uri Redirecting number; ?Privacy=history; cause=**302**; index=1.1, hi-targeted-to-uri called party number; cause= **Cause_value**; index=1.1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion" Redirection counter = 2 Redirecting indicator = 4 Redirecting reason = ISUP_RR Original called number Presentation restriction: Presentation restricted Redirecting number Presentation restriction: Presentation restricted	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC 200 OK BYE ← ←	

IAM		**INVITE**	
ISUP Parameter or	**Source value of parameter field**	**SIP component**	**Derived value of header field**
Redirection Information ISUP_RR	**Redirecting reason**	**History-Info header**	**Cause parameter**
	unknown '0000'B	**Cause_value**	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609011	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*IAM with Original call number and redirecting number received, Redirection counter value 3* Ensure that the SUT if the IAM is received with **Redirecting number, original called number** and **redirection information**, the redirection counter value is "3", an INVITE is sent containing a History-Info header. The Original called number is contained in the hi-tatgeted-to uri in the index 1. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1.1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.	
SIP Parameter values:	INVITE: History-Info header hi-targeted-to-uri Original called number; index=1, hi-targeted-to-uri Dummy entry(PIXIT); cause=**302**; index=1.1, hi-targeted-to-uri Redirecting number; cause=**486**; index=1.1.1, hi-targeted-to-uri called party number; cause= **Cause_value**; index=1.1.1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion" Redirection counter = 3 Redirecting reason = ISUP_RR Original called number Redirecting number	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC 200 OK BYE ← ←	

IAM		**INVITE**	
ISUP Parameter or	**Source value of parameter field**	**SIP component**	**Derived value of header field**
Redirection Information ISUP_RR	**Redirecting reason**	**History-Info header**	**Cause parameter**
	unknown '0000'B	**Cause_value**	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

TP609012	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
ISUP selection criteria:		
Test purpose:	*IAM with Original call number and redirecting number Presentation allowed received, Redirection counter value 2, Redirecting indicator indicates "all redirection information presentation restricted"* Ensure that the SUT if the IAM is received with **Redirecting number, original called number** Presentation allowed and **redirection information** Presentation restricted, the redirection counter value is "2", an INVITE is sent containing a History-Info header. The Original called number is contained in the hi-tatgeted-to uri in the index 1. The Redirecting number is contained in the in the hi-targeted-to-uri in index 1.1, the called party number is contained in the in the hi-targeted-to-uri in index 1.1.1. The cause parameter value in the latest history entry is mapped into the redirection reason indicator.	
SIP Parameter values:	INVITE: History-Info header hi-targeted-to-uri Original called number?Privacy=history; index=1, hi-targeted-to-uri Redirecting number?Privacy=history; cause=**302**; index=1.1, hi-targeted-to-uri called party number; cause= **Cause_value**; index=1.1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion" Redirection counter = 2 Redirecting indicator = 4 Redirecting reason = ISUP_RR Original called number Presentation restriction: Presentation allowed Redirecting number Presentation restriction: Presentation allowed	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC 200 OK BYE ← ←	

IAM		**INVITE**	
ISUP Parameter or	**Source value of parameter field**	**SIP component**	**Derived value of header field**
Redirection Information	**Redirecting reason**	**History-Info header**	**Cause parameter**
ISUP_RR	unknown '0000'B	**Cause_value**	404
	Unconditional '0011'B		302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not reachable		503

<table>
<tr><th>TP609013</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.5.4</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/SS/CDIV/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>181 Received, Notification subscription option according the Privacy header in the History-Info header</i>

Ensure that the SUT (when no ACM has been sent before) on receipt of 181 (Call Is Being Forwarded) containing the History-Index, Privacy, priv-value component and the History-Info header, Privacy, priv-value component concerning the diverted-to uri

Sends an ACM message indicating a first diversion with the Backward call indicators parameter coded
Called party's status indicator = no indication
Redirection number:
Redirection number: derived from the Hi-target-to-uri of the last History-Info entry
the Call diversion information parameter
Notification subscription option = ISUP_NSO
and the Generic notification indicator parameter = call is diverting</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">181: History-Info header
hi-targeted-to-uri Redirecting number; index=1,
hi-targeted-to uri diverted to user; cause=Cause value?Privacy=priv-value; index=1.1</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">ACM
BCI: No indication (00),
GenNot: Call is diverting (1111011),
Call diversion Info: ISUP_NSO
Redirection number: derived from the Hi-target-to-uri of the last History-Info entry</td></tr>
<tr><td>Comments:</td><td colspan="2">ISUP SUT SIP
IAM → → INVITE
ACM ← ← 181 Being Forwarded
CPG ← ← 180 Ringing
ANM ← ← 200 OK INVITE
→ ACK
Communication
REL → → BYE
RLC ← ← 200 OK BYE</td></tr>
</table>

	SIP component **History-Info header,** **priv-value component**	**Call diversion information *Notification subscription options*** **ISUP_NSO**
VA_01	Privacy header field absent	ISUP_NSO = presentation allowed with redirection number
VA_02	Privacy "none"	ISUP_NSO = presentation allowed with redirection number
VA_03	Privacy "history"	ISUP_NSO = presentation allowed without redirection number

TP609014	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*181 received, ACM no indication is sent: NSO Presentation not allowed* Ensure that the SUT (when no ACM has been sent before) on receipt of 181 (Call Is Being Forwarded) containing priv-value history is set to the hist-info element concerning the redirecting uri and the diverted-to-uri then Sends of an **ACM** message indicating a first diversion with the **Backward call indicators parameter** coded Called party's status indicator = no indication **Redirection number:** Redirection number: derived from the Hi-target-to-uri of the last History-Info entry the **Call diversion information parameter** Notification subscription option = *presentation not allowed* and the **Generic notification indicator parameter** = call is diverting	
SIP Parameter values:	181: History-Info header hi-targeted-to-uri Redirecting number?**Privacy=history**; index=1, hi-targeted-to uri diverted to user; cause=Cause value?**Privacy=history**; index=1.1	
ISUP Parameter values:	ACM BCI: No indication (00), GenNot: Call is diverting (1111011), Call diversion Info: *presentation not allowed* Redirection number: derived from the Hi-target-to-uri of the last History-Info entry	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 181 Being Forwarded CPG ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK REL → → BYE RLC ← ← 200 OK BYE	

TP609015	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*181 received sending of Redirection number restriction parameter in the ACM* Ensure that the SUT (when no ACM has been sent before) on receipt of 181 (Call Is Being Forwarded) containing the History-Info header, Privacy, priv-value component Sends of an ACM message indicating a first diversion with the Backward call indicators parameter coded Called party's status indicator = no indication **Redirection number:** Redirection number: derived from the Hi-target-to-uri of the last History-Info entry Redirection number restriction indicator Redirection number restriction option = ISUP_ReNrReIn and the Generic notification indicator parameter = call is diverting	
SIP Parameter values:	181: History-Info header hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value?Privacy=**priv-value**; index=1.1	
ISUP Parameter values:	ACM: BCI: No indication (00), GenNot: Call is diverting (1111011), Redirection number: derived from the Hi-target-to-uri of the last History-Info entry Redirection number restriction indicator: ISUP_ReNrReIn	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 181 Being Forwarded CPG ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK REL → → BYE RLC ← ← 200 OK BYE	

	History-Info header Privacy, priv-value component	**Redirection number restriction indicator ISUP_ReNrReIn**
VA_01	Privacy "history"	Presentation restricted
VA_02	Privacy header field absent	Presentation allowed or absent
VA_03	Privacy "none"	Presentation allowed or absent

TP609016	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*181 received, coding of notification subscription option in the CPG Progress* Ensure that the SUT, when an ACM has been sent before, on receipt of 181 (Call Is Being Forwarded) containing the History-Index, Privacy, priv-value component and the History-Info header, Privacy, priv-value component concerning the diverted-to uri Sends a CPG message indicating a first diversion with the **Event information parameter** coded Event indicator = PROGRESS, **Redirection number:** Redirection number: derived from the Hi-target-to-uri of the last History-Info entry the **Call diversion information parameter** Notification subscription option = ISUP_NSO and the **Generic notification indicator parameter** = call is diverting	
SIP Parameter values:	181: History-Info header hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value?Privacy=**Priv-value**; index=1.1	
ISUP Parameter values:	CPG: Event indicator = PROGRESS, GenNot: Call is diverting (1111011), Call diversion Info: ISUP_NSO Redirection number: derived from the Hi-target-to-uri of the last History-Info entry	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing CPG ← ← 181 Being Forwarded ANM ← ← 200 OK INVITE → ACK REL → → BYE RLC ← ← 200 OK BYE	

	SIP component **History-Index Privacy,** **priv-value component**	**Call diversion information *Notification subscription options*** **ISUP_NSO**
VA_01	Privacy header field absent	ISUP_NSO = presentation allowed with redirection number
VA_02	Privacy "none"	ISUP_NSO = presentation allowed with redirection number
VA_03	Privacy "history"	ISUP_NSO = presentation not allowed,

TP609017	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*181 received Privacy=history concerning the redirecting and the diverted-to URI setting of NSO in the CPG Progress* Ensure that the SUT, when an ACM has been sent before, on receipt of 181 (Call Is Being Forwarded) containing priv-value history is set to the hist-info element concerning the redirecting uri and the diverted-to-uri then Sends a CPG message indicating a first diversion with the **Event information parameter** coded Event indicator = PROGRESS, **Redirection number:** Redirection number: derived from the Hi-target-to-uri of the last History-Info entry the **Call diversion information parameter** Notification subscription option = *presentation not allowed* and the **Generic notification indicator parameter** coded Notification indicator = call is diverting,	
SIP Parameter values:	181: History-Info header hi-targeted-to-uri Redirecting number?**Privacy=history**; index=1, hi-targeted-to uri diverted to user; cause=Cause value?**Privacy=history**; index=1.1	
ISUP Parameter values:	CPG: Event indicator = PROGRESS, GenNot: Call is diverting (1111011), Call diversion Info: *presentation not allowed*	
Comments:		

ISUP		**SUT**		**SIP**
IAM	→		→	INVITE
ACM	←		←	180 Ringing
CPG	←		←	181 Being Forwarded
ANM	←		←	200 OK INVITE
			→	ACK
		Communication		
REL	→		→	BYE
RLC	←		←	200 OK BYE

TP092018	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*181 received setting of Redirection number restriction in the CPG Progress* Ensure that the SUT, when an ACM has been sent before, on receipt of 181 (Call Is Being Forwarded) **containing the** History-Info header, Privacy, priv-value component Sends a CPG message indicating a first diversion with the **Event information parameter** coded Event indicator = PROGRESS, **Redirection number:** Redirection number: derived from the Hi-target-to-uri of the last History-Info entry **Redirection number restriction indicator**= ISUP_ReNrReIn and the **Generic notification indicator parameter** = call is diverting	
SIP Parameter values:	181: History-Info header hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value?Privacy=**priv-value**; index=1.1	
ISUP Parameter values:	CPG: Event indicator = PROGRESS, GenNot: Call is diverting (1111011), Redirection number: ISUP_ReNr Redirection number restriction indicator: ISUP_ReNrReIn	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing CPG ← ← 181 Being Forwarded ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC ← ← 200 OK BYE	

	History-Info header Privacy, priv-value component	**Redirection number restriction indicator ISUP_ReNrReIn**
VA_01	Privacy "history"	Presentation restricted
VA_02	Privacy "none"	Presentation allowed or absent
VA_03	Privacy header field absent	Presentation allowed or absent

TP609019	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*180 received, CPG Alerting is sent, setting of NSO.* **Ensure that the SUT on receipt of 180 (Ringing)** (181 Call Is Being Forwarded was received before) **containing the** History-Info header, Privacy, priv-value component concerning diverted-to uri Sends a CPG message indicating a first diversion with the **Event information parameter** coded Event indicator = ALERTING, **Redirection number:** Redirection number: derived from the Hi-target-to-uri of the last History-Info entry Notification subscription option = **ISUP_NSO** and the **Generic notification indicator parameter** = call is diverting	
SIP Parameter values:	180: History-Info header hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value?Privacy=**priv-value**; index=1.1	
ISUP Parameter values:	CPG: Event indicator = ALERTING, GenNot: Call is diverting (1111011), Redirection number: derived from the Hi-target-to-uri of the last History-Info entry Notification subscription option: ISUP_NSO	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 181 Being Forwarded CPG ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC ← ← 200 OK BYE	

	SIP component **History-Info header,** **priv-value component**	**Call diversion information *Notification subscription options*** **ISUP_NSO**
VA_01	Privacy header field absent	ISUP_NSO = presentation allowed with redirection number
VA_02	Privacy "none"	ISUP_NSO = presentation allowed with redirection number
VA_03	Privacy "history"	ISUP_NSO = presentation allowed without redirection number

TP609020	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*180 received, CPG Alerting is sent, setting of NSO.* **Ensure that the SUT on receipt of 180 (Ringing)** (181 Call Is Being Forwarded was received before) **containing the** History-Info header, Privacy, priv-value component concerning redirecting and diverted-to uri Sends a CPG message indicating a first diversion with the **Event information parameter** coded Event indicator = ALERTING, **Redirection number:** Notification subscription option = *presentation not allowed* Redirection number: derived from the Hi-target-to-uri of the last History-Info entry and the **Generic notification indicator parameter** coded Notification indicator = call is diverting	
SIP Parameter values:	180: History-Info header hi-targeted-to-uri Redirecting number?**Privacy=history**; index=1, hi-targeted-to uri diverted to user; cause=Cause value?**Privacy=history**; index=1.1	
ISUP Parameter values:	CPG: Event indicator = ALERTING, GenNot: Call is diverting (1111011), Redirection number: derived from the Hi-target-to-uri of the last History-Info entry Notification subscription option: Presentation not allowed	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 181 Being Forwarded CPG ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC ← ← 200 OK BYE	

TP609021	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*180 received, CPG Alerting is sent, setting of Redirection number restriction.* **Ensure that the SUT on receipt of 180 (Ringing)** (181 Call Is Being Forwarded was received before) **containing the** History-Info header, Privacy, priv-value component concerning diverted-to uri Sends a CPG message indicating a first diversion with the **Event information parameter** coded Event indicator = ALERTING, **Redirection number:** Redirection number restriction indicator = ISUP_RNR and the Generic notification indicator parameter coded Notification indicator = call is diverting	
SIP Parameter values:	180: History-Info header hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value?Privacy=**priv-value**; index=1.1	
ISUP Parameter values:	CPG: Event indicator = ALERTING, Redirection number restriction indicator: ISUP_ReNrReIn	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 181 Being Forwarded CPG ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC ← ← 200 OK BYE	

	History-Info header Privacy, priv-value component	**Redirection number restriction indicator ISUP_ReNrReIn**
VA_01	Privacy "history"	Presentation restricted
VA_02	Privacy "none"	Presentation allowed or absent
VA_03	Privacy header field absent	Presentation allowed or absent

TP609022	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*Redirection number restriction in ANM* Ensure that the SUT on receipt of 200 (OK) containing the History-Info header, Privacy, priv-value component Sends a ANM message with Redirection number restriction indicator: ISUP_ReNrReIn	
SIP Parameter values:	200: History-Info header hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value?Privacy**=priv-value**; index=1.1	
ISUP Parameter values:	ANM: Redirection number restriction indicator: ISUP_ReNrReIn	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 181 Being Forwarded CPG ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK REL → → BYE RLC ← ← 200 OK BYE	

	History-Info header Privacy, priv-value component	**Redirection number restriction indicator ISUP_ReNrReIn**
VA_01	Privacy "history"	Presentation restricted
VA_02	Privacy "none"	Presentation allowed or absent
VA_03	Privacy header field absent	Presentation allowed or absent

TP609023	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*181 Received, no mapping to an ACM* Ensure that the SUT (when no ACM has been sent before) on receipt of 181 (Call Is Being Forwarded) containing the History-Index no ACM is sent	
SIP Parameter values:	181: History-Info header hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value; index=1.1	
ISUP Parameter values:		
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ← 181 Being Forwarded ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC ← ← 200 OK BYE	

TP609024	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15	
Test purpose:	*181 received, not mapped to a CPG* Ensure that the SUT, when an ACM has been sent before, on receipt of 181 (Call Is Being Forwarded) containing the History-Index, no CPG is sent	
SIP Parameter values:	181: History-Info header hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value?Privacy=**Priv-value**; index=1.1	
ISUP Parameter values:		
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing ← 181 Being Forwarded ANM ← ← 200 OK INVITE → ACK REL → → BYE RLC ← ← 200 OK BYE	

TP609025	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 5/12 AND NOT PICS 5/13 AND NOT PICS 5/14 AND NOT PICS 5/15	
Test purpose:	*180 received, no mapping.* **Ensure that the SUT on receipt of 180 (Ringing)** (181 Call Is Being Forwarded was received before) **containing the** History-Info header, the History-Info header is not mapped	
SIP Parameter values:	180: History-Info header hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value?Privacy=**priv-value**; index=1.1	
ISUP Parameter values:		
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ← 181 Being Forwarded ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK **Communication** REL → → BYE RLC ← ← 200 OK BYE	

TP609026	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.5.4**
TSS reference:	ISUP-SIP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 5/12 AND NOT PICS 5/13 AND NOT PICS 5/14 AND NOT PICS 5/15	
Test purpose:	*No mapping of History-Info header in the 200 OK INVITE* Ensure that the SUT on receipt of 200 (OK) containing the History-Info header, Privacy, priv-value component the History-Info header is not mapped	
SIP Parameter values:	200: History-Info header hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value?Privacy**=priv-value**; index=1.1	
ISUP Parameter values:	No mapping	
Comments:	**ISUP** **SUT** **SIP** IAM → → INVITE ← 181 Being Forwarded ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK REL → → BYE RLC ← ← 200 OK BYE	

6.3.2.10 User to user signalling (UUS)

TP610001	**SIP reference: RFC 3261 [6]**	**ISUP reference: ITU-T Rec Q.1912.5 [32], annex B.21 ITU-T Rec Q.737.1 [33], clause 1.1.7**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*User-to-user service 1 implicit request not supported, User-to-user information discarded by the network* Ensure that the SUT if the IAM is received with **User-to-user information as an implicit service 1 request** returns a User-to-user indicator in the ACM "UUI discarded by the network" and continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: User-to-indicator "UUI discarded by the network", Service 1 response "No indication".	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610002	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737 [33], clause 1.1.7**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	*User-to-user service 1 explicit request not essential not supported, service not provided response* Ensure that the SUT if the IAM is received with an **explicit service 1 request "Not essential"** returns a User-to-user indicator in the ACM "Service 1 not provided" and continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: User-to-indicator Service 1 response "Not provided"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33], clause 1.1.7**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	*User-to-user service 1 explicit request essential not supported, rejected by sending a REL* Ensure that the SUT if the IAM is received with an **explicit service 1 request** "essential" returns a REL with cause #29 and an diagnostics containing the user-to-user indicator parameter name.	
SIP Parameter values:	No action	
ISUP Parameter values:	REL: cause #29, diagnostics value 0x2a	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → REL #29 ← RLC →	

TP610004	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737 [33], clause 1.2.7**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	*User-to-user service 2 explicit request not essential not supported, service not provided response* Ensure that the SUT if the IAM is received with an **explicit service 2 request** "Not essential" returns a User-to-user indicator in the ACM "Service 2 not provided" and continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: User-to-indicator Service 2 response "Not provided"	
Comments:		

ISUP/BICC		SUT		SIP
IAM	→		→	INVITE
ACM	←		←	180 Ringing
	Ringing tone			
ANM	←		←	200 OK INVITE
			→	ACK
		Conversation		
REL	→		→	BYE
RLC	←		←	200 OK BYE

TP610005	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33], clause 1.2.7**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	*User-to-user service 2 explicit request essential not supported, rejected by sending a REL* Ensure that the SUT if the IAM is received with an **explicit service 2 request** "essential" returns a REL with cause #29 and an diagnostics containing the user-to-user indicator parameter name.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	REL: cause #29, diagnostics value 0x2a	
Comments:		

ISUP/BICC		SUT		SIP
IAM	→			
REL #29	←			
RLC	→			

TP610006	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33], clause 1.3.7.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	*User-to-user service 3 explicit request not essential not supported, service not provided response* Ensure that the SUT if the IAM is received with an **explicit service 3 request** "Not essential" returns a User-to-user indicator in the ACM "Service 3 not provided" and continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: User-to-indicator, Service 3 response "Not provided"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610007	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33], clause 1.3.7.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	*User-to-user service 3 explicit request essential not supported, rejected by sending a REL* Ensure that the SUT if the IAM is received with an **explicit service 3 request** "essential" returns a REL with cause #29 and an diagnostics containing the user-to-user indicator parameter name.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	REL: cause #29, diagnostics value 0x2a	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → REL #29 ← RLC →	

TP610008	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33], clause 1.3.7.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	*User-to-user service 3 explicit request not essential not supported in the confirmed state, rejected by sending a FRJ* Ensure that the SUT if the FAR is received with an **explicit service 3 request** "Not essential" returns a FRJ with cause #29.	
SIP Parameter values:	No action	
ISUP Parameter values:	FRJ: User-to-user indicator = "Service 3 not provided"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation FAR → FRJ ← Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610009	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33], clause 1.1.5.2.5.2.2**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	*User-to-user service 1 explicit request not essential not supported, no response* Ensure that the SUT if the IAM is received with an **explicit service 1 request** "Not essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No mapping	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610010	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33],** **clause 1.1.5.2.5.2.2**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	*User-to-user service 1 explicit request essential not supported, no response* Ensure that the SUT if the IAM is received with an **explicit service 1 request** "essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No action	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610011	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33],** **clause 1.2.5.2.5.2.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	*User-to-user service 2 explicit request not essential not supported, no response* Ensure that the SUT if the IAM is received with an **explicit service 2 request** "Not essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No mapping	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610012	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33],** **clause 1.2.5.2.5.2.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	*User-to-user service 2 explicit request essential not supported, no response* Ensure that the SUT if the IAM is received with an explicit **service 2 request** "essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No action	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610013	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33],** **clause 1.3.5.2.5.2.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	*User-to-user service 3 explicit request not essential not supported, no response* Ensure that the SUT if the IAM is received with an **explicit service 3 request** "Not essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No mapping	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610014	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33],** **clause 1.3.5.2.5.2.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	*User-to-user service 3 explicit request essential not supported, no response* Ensure that the SUT if the IAM is received with an **explicit service 3 request** "essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No action	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610015	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33],** **clause 1.3.5.2.5.2.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/1 OR NOT PICS 11/3	
Test purpose:	*User-to-user service 3 explicit request not essential not supported in the confirmed state, no response* Ensure that the SUT if the FAR is received with an **explicit service 3 request** "Not essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No action	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation FAR → REL → → BYE RLC ← ← 200 OK BYE	

TP610016	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33],** **clause 1.3.5.2.5.2.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*User-to-user service 1 implicit request is mapped in the User-to-User header field in the INVITE request* Ensure that the SUT if the IAM contains a User-to-user information parameter, a User-to-User header is included in the INVITE request and the uuidata component is derived from the User-to-user information.	
SIP Parameter values:	INVITE: User-to-User: uuidata derived from the User-to-user information	
ISUP Parameter values:	IAM: User-to-user information (PIXIT)	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610017	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33],** **clause 1.3.5.2.5.2.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*User-to-user service 1 implicit response is mapped in the* User-to-user information parameter *in the ACM* Ensure that the SUT if the 180 Ringing contains a User-to-User header, a User-to-user information parameter is included in the ACM and the the User-to-user information is derived from the uuidata component. User-to-User header starts with the first octet being the protocol discriminator and followed by the user information octets	
SIP Parameter values:	180: User-to-User: uuidata derived from the User-to-user information (PIXIT)	
ISUP Parameter values:	ACM: User-to-user information	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610018	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33],** **clause 1.3.5.2.5.2.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*User-to-user service 1 implicit response is mapped in the* User-to-user information parameter *in the ANM* Ensure that the SUT if the 200 OK INVITE contains a User-to-User header, a User-to-user information parameter is included in the ANM and the the User-to-user information is derived from the uuidata component. User-to-User header starts with the first octet being the protocol discriminator and followed by the user information octets	
SIP Parameter values:	200: User-to-User: uuidata derived from the User-to-user information (PIXIT)	
ISUP Parameter values:	ANM: User-to-user information	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP610019	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ITU-T Rec Q.1912.5 [32], annex B.21** **ITU-T Rec Q.737.1 [33],** **clause 1.3.5.2.5.2.1**
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	*User-to-user service 1 implicit response is mapped in the* User-to-user information parameter *in the REL* Ensure that the SUT if the BYE contains a User-to-User header, a User-to-user information parameter is included in the REL and the the User-to-user information is derived from the uuidata component. User-to-User header starts with the first octet being the protocol discriminator and followed by the user information octets	
SIP Parameter values:	BYE: User-to-User: uuidata derived from the User-to-user information (PIXIT)	
ISUP Parameter values:	REL: User-to-user information	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation REL ← ← BYE RLC → → 200 OK BYE	

6.3.2.11 Explicit call transfer (ECT)

TP611001	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.8**
TSS reference:	ISUP-SIP/SS/ECT/	
SIP selection criteria:		
ISUP selection criteria:	PICS 12/1 AND NOT PICS 13/3	
Test purpose:	*Loop prevention procedure supported, interworking of "call transfer" indication not supported* Ensure that the SUT if a LOP(request) is received returns a LOP (response) with the indication "insufficient information" continue without disrupting the SIP signalling procedure. Ensure that the SUT if a FAC is received continue without disrupting the SIP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	LOP: Response "insufficient information"	

Comments:

ISUP/BICC		**SUT**		**SIP**
IAM	→		→	INVITE
ACM	←		←	180 Ringing
	Ringing tone			
ANM	←		←	200 OK INVITE
			→	ACK
		Conversation		
LOP	→			
LOP	←			
FAC(Call transfer active)	→			
		Conversation		
REL	→		→	BYE
RLC	←		←	200 OK BYE

TP611002	**SIP reference: RFC 3261 [6]**	**ISUP reference: ES 283 027 [1], clause 7.4.8**
TSS reference:	ISUP-SIP/SS/ECT/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 12/1 AND NOT PICS 13/3	
Test purpose:	*Loop prevention procedure not supported, interworking of "call transfer" indication not supported* Ensure that the SUT if a LOP(request) is received continue without disrupting the SIP signalling procedure. Ensure that the SUT if a FAC is received continue without disrupting the SIP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:		
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation LOP → FAC(Call transfer active) → Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP611003	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.8**
TSS reference:	ISUP-SIP/SS/ECT/	
SIP selection criteria:		
ISUP selection criteria:	PICS 12/1 AND PICS 13/3	
Test purpose:	*Loop prevention procedure supported, interworking of "call transfer" indication in FAC supported* Ensure that the SUT if a LOP(request) is received returns a LOP (response) with the indication "insufficient information" continue without disrupting the SIP signalling procedure. Ensure that the SUT if a FAC is received an INVITE is sent and the SDP contains an a-line set to "sendrecv".	
SIP Parameter values:	Re-INVITE SDP a=sendrecv	
ISUP Parameter values:	LOP: Response "insufficient information" FAC: Generic notification = "call transfer, active"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation CPG(hold) → INVITE(sendonly) ← 200 OK INVITE(recvonly) → ACK LOP → LOP ← FAC(Call transfer, active) → → INVITE(sendrecv) ← 200 OK INVITE(sendrecv) → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

TP611004	**SIP reference: RFC 3261 [6]**	**ISUP reference:** **ES 283 027 [1], clause 7.4.8**
TSS reference:	ISUP-SIP/SS/ECT/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 12/1 AND PICS 13/3	
Test purpose:	*Loop prevention procedure supported, interworking of "call transfer" indication in FAC supported* Ensure that the SUT if a FAC is received an INVITE is sent and the SDP contains an a-line set to "sendrecv".	
SIP Parameter values:	Re-INVITE SDP a=sendrecv	
ISUP Parameter values:	FAC: Generic notification = "call transfer, active"	
Comments:	**ISUP/BICC** **SUT** **SIP** IAM → → INVITE ACM ← ← 180 Ringing Ringing tone ANM ← ← 200 OK INVITE → ACK Conversation CPG(hold) → INVITE(sendonly) ← 200 OK INVITE(recvonly) → ACK LOP → FAC(Call transfer active) → → INVITE(sendrecv) ← 200 OK INVITE(sendrecv) → ACK Conversation REL → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP611005</th><th>SIP reference: RFC 3261 [6]</th><th>ISUP reference:
ES 283 027 [1], clause 7.4.8</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/SS/ECT/</td></tr>
<tr><td>SIP selection criteria:</td><td colspan="2"></td></tr>
<tr><td>ISUP selection criteria:</td><td colspan="2">PICS 13/3</td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Interworking of "call transfer" indication in CPG supported</i>

Ensure that the SUT if a CPG Generic notification "call transfer, active" is received an INVITE is sent and the SDP contains an a-line set to "sendrecv".</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">Re-INVITE SDP a=sendrecv</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">CPG: Generic notification = "call transfer, active"</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>ISUP/BICC</th><th></th><th>SUT</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td></td><td>Ringing tone</td><td></td><td></td><td></td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>CPG(hold)</td><td></td><td></td><td>→</td><td>INVITE(sendonly)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(recvonly)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td>LOP</td><td>→</td><td></td><td></td><td></td></tr>
<tr><td>CPG(Call transfer active)</td><td>→</td><td></td><td>→</td><td>INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INVITE(sendrecv)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td></td><td>Conversation</td><td></td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>
</td></tr>
</table>

6.3.2.12 Anonymous Call Rejection (ACR)

<table>
<tr><th>TP612001</th><th>ACR-CB Reference:
4.7.1.3.1</th><th>Selection criteria:</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/SS/ACR</td></tr>
<tr><td>Preconditions:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Mapping of 433 Anonymity Disallowed to REL cause 24</i>

Ensure that the 433 Anonymity Disallowed final response received to due the ACR service is mapped into a REL cause 24 "<i>call rejected due to ACR supplementary service</i>"</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">433 Anonymity Disallowed</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">REL cause value 24 "<i>call rejected due to ACR supplementary service</i>"</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>ISUP</th><th></th><th>MGCF</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>REL(24)</td><td>←</td><td></td><td>←</td><td>433 Anonymity Disallowed</td></tr>
<tr><td>RLC</td><td>→</td><td></td><td>→</td><td>ACK</td></tr>
</table>
</td></tr>
</table>

TP612002	**ACR-CB Reference:** **4.7.1.3.1**	**Selection criteria:**
TSS reference:	ISUP-SIP/SS/ACR	
Preconditions:		
Test purpose:	*Mapping of 603 Decline to REL cause 21* Ensure that the 603 Decline final response received to due the ACR service is mapped into a REL cause 21 "*call rejected*"	
SIP Parameter values:	603 Decline	
ISUP Parameter values:	REL cause value 21 "*call rejected*"	
Comments:	ISUP MGCF SIP IAM → → INVITE ← 100 Trying REL(24) ← ← 603 Decline RLC → → ACK	

6.3.2.13 Call waiting (CW)

FFS

6.3.2.14 Malicious call identification (MCID)

TP614001	**MCID Reference:** **4.7.1.2**	**Selection criteria:** PICS 1/6
TSS reference:	ISUP-SIP/SS/MCID/	
Preconditions:		
Test purpose:	*Mapping of XML mcid request (McidRequestIndicator)* Ensure that the XML mcid McidRequestIndicator contained in a received INFO request mapped into the MCID request indicator requested in the sent IDR	
SIP Parameter values:	INFO XML mcid request McidRequestIndicator = "1"	
ISUP Parameter values:	IDR: MCID request indicator: MCID requested	
Comments:	**ISUP** **MGCF** **SIP** IAM → → INVITE ← 100 Trying IDR(MCID request indicator) ← ← INFO (XML mcid request) → 200 OK INFO ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK Communication REL → → BYE RLC ← ← 200 OK BYE	

TP614002	**MCID Reference:** **4.7.1.2**	**Selection criteria:** **PICS 1/6 AND PICS 1/7**
TSS reference:	ISUP-SIP/SS/MCID/	
Preconditions:		
Test purpose:	*Mapping of XML mcid request (HoldingIndicator)* Ensure that the XML mcid HoldingIndicator is mapped into the MCID request indicator holding requested in the sent IDR	
SIP Parameter values:	INFO: XML mcid request HoldingIndicator = "1"	
ISUP Parameter values:	IDR: *Holding indicator (national use)*: holding requested	
Comments:	**ISUP** **MGCF** **SIP** IAM → → INVITE ← 100 Trying IDR(MCID request indicator) ← ← INFO (XML mcid request) → 200 OK INFO ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK Communication REL → → BYE RLC ← ← 200 OK BYE	

TP614003	**MCID Reference:** **4.7.1.2**	**Selection criteria:** **PICS 1/6**
TSS reference:	ISUP-SIP/SS/MCID/	
Preconditions:		
Test purpose:	*Mapping of IRS (McidResponseIndicator)* Ensure that MCID response indicator provided, contained in an IRS is mapped into the XML mcid response McidResponseIndicator.	
SIP Parameter values:	INFO: XML mcid request McidRequestIndicator = "1" INFO: XML mcid response McidResponseIndicator = "1"	
ISUP Parameter values:	IDR: MCID request indicator: MCID requested IRS: MCID response indicator: MCID provided	
Comments:	**ISUP** **MGCF** **SIP** IAM → → INVITE ← 100 Trying IDR(MCID request indicator) ← ← INFO (XML mcid request) → 200 OK INFO IRS (MCID response indicator) → → INFO (XML mcid response) ← 200 OK INFO ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE → ACK Communication REL → → BYE RLC ← ← 200 OK BYE	

<table>
<tr><th>TP614004</th><th>MCID Reference:
4.7.1.2</th><th>Selection criteria:
PICS 1/6 AND NOT PICS 1/7</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/SS/MCID/</td></tr>
<tr><td>Preconditions:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Mapping of IRS (HoldingProvidedIndicator)</i>
Ensure that MCID response indicator holding provided, contained in an IRS is mapped into the XML mcid response HoldingProvidedIndicator.</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INFO:
XML mcid
request
HoldingIndicator = "1"
INFO:
XML mcid
response
HoldingProvidedIndicator = "0"</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IDR: <i>Holding indicator (national use)</i>:
IRS: <i>Hold provided indicator (national use)</i></td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>ISUP</th><th></th><th>MGCF</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>➔</td><td></td><td>➔</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>🡐</td><td>100 Trying</td></tr>
<tr><td>IDR(MCID request indicator)</td><td>🡐</td><td></td><td>🡐</td><td>INFO (XML mcid request)</td></tr>
<tr><td></td><td></td><td></td><td>➔</td><td>200 OK INFO</td></tr>
<tr><td>IRS (no MCID response indicator)</td><td></td><td></td><td>➔</td><td>INFO (XML mcid response)</td></tr>
<tr><td></td><td></td><td></td><td>🡐</td><td>200 OK INFO</td></tr>
<tr><td>ACM</td><td>🡐</td><td></td><td>🡐</td><td>180 Ringing</td></tr>
<tr><td>ANM</td><td>🡐</td><td></td><td>🡐</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>➔</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>REL</td><td>➔</td><td></td><td>➔</td><td>BYE</td></tr>
<tr><td>RLC</td><td>🡐</td><td></td><td>🡐</td><td>200 OK BYE</td></tr>
</table>
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<table>
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4.7.1.2</th><th>Selection criteria:
PICS 1/6 AND PICS 1/7</th></tr>
<tr><td>TSS reference:</td><td colspan="2">ISUP-SIP/SS/MCID/</td></tr>
<tr><td>Preconditions:</td><td colspan="2"></td></tr>
<tr><td>Test purpose:</td><td colspan="2"><i>Mapping of IRS (HoldingProvidedIndicator)</i>
Ensure that MCID response indicator holding provided, contained in an IRS is mapped into the XML mcid response HoldingProvidedIndicator (Holding indicator is not for national use).</td></tr>
<tr><td>SIP Parameter values:</td><td colspan="2">INFO:
XML mcid
request
HoldingIndicator = "1"
INFO:
XML mcid
response
HoldingProvidedIndicator = "1"</td></tr>
<tr><td>ISUP Parameter values:</td><td colspan="2">IDR: <i>Holding indicator</i>: holding requested
IRS: <i>Hold provided indicator</i> holding provided</td></tr>
<tr><td>Comments:</td><td colspan="2">
<table>
<tr><th>ISUP</th><th></th><th>MGCF</th><th></th><th>SIP</th></tr>
<tr><td>IAM</td><td>→</td><td></td><td>→</td><td>INVITE</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>100 Trying</td></tr>
<tr><td>IDR(MCID request indicator)</td><td>←</td><td></td><td>←</td><td>INFO (XML mcid request)</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>200 OK INFO</td></tr>
<tr><td>IRS (no MCID response indicator)</td><td></td><td></td><td>→</td><td>INFO (XML mcid response)</td></tr>
<tr><td></td><td></td><td></td><td>←</td><td>200 OK INFO</td></tr>
<tr><td>ACM</td><td>←</td><td></td><td>←</td><td>180 Ringing</td></tr>
<tr><td>ANM</td><td>←</td><td></td><td>←</td><td>200 OK INVITE</td></tr>
<tr><td></td><td></td><td></td><td>→</td><td>ACK</td></tr>
<tr><td></td><td colspan="3">Communication</td><td></td></tr>
<tr><td>REL</td><td>→</td><td></td><td>→</td><td>BYE</td></tr>
<tr><td>RLC</td><td>←</td><td></td><td>←</td><td>200 OK BYE</td></tr>
</table>
</td></tr>
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Annex A (informative): Bibliography

- ITU-T Recommendation Q.731.7 (1997): "Stage 3 description for number identification supplementary services using Signalling System No. 7: Malicious call identification (MCID)".
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History

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