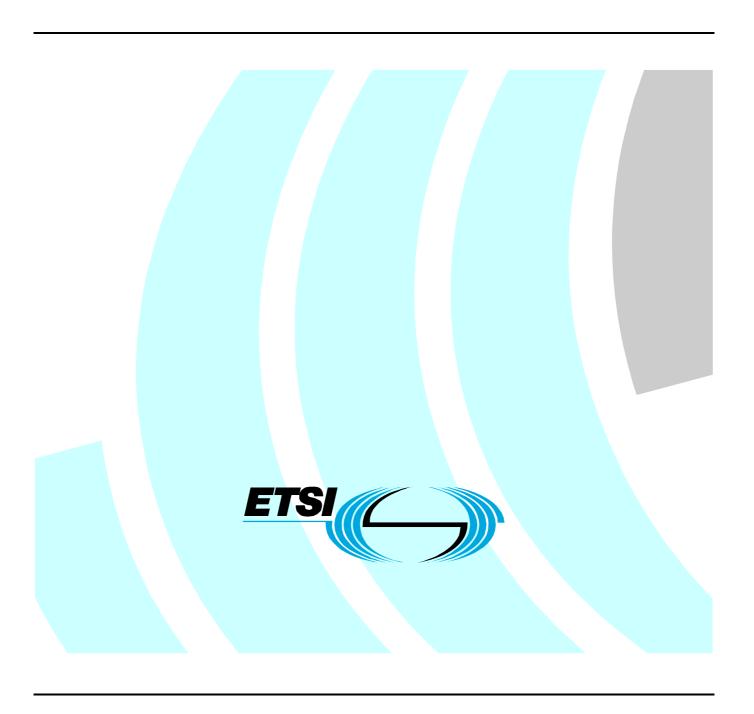
# ETSITS 186 009-2 V2.2.1 (2011-03)

Technical Specification

Technical Committee for IMS Network Testing (INT);
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

#### RTS/INT-00022-2

#### Keywords

BICC, interworking, SIP, testing, TSS&TP

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#### **Foreword**

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

The present document is part 2 of a multi-part deliverable covering SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks, as identified below:

- 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)".

# 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 identical.

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

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

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#### 2.1 Normative references

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

[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] Void.
- [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; ISDN User Part (ISUP) version 2 for the international interface; Part 1: Basic services [ITU-T Recommendations Q.761 to Q.764 (1993), 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".

- [36] ITU-T Recommendation Q.767: "Application of the ISDN User Part of CCITT signalling system No. 7 for international ISDN interconnections".
- [37] ITU-T Recommendation G.711 (1988): "Pulse code modulation (PCM) of voice frequencies".

### 2.2 Informative references

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

- [i.1] ITU-T Recommendation T.101 (1994): "International interworking for Videotex services".
- [i.2] ITU-T Recommendation T.102 (1993): "Syntax-based Videotex end-to-end protocols for the circuit mode ISDN".
- [i.3] ITU-T Recommendation X.200 (1994): "Information technology Open Systems Interconnection Basic Reference Model: The basic model".
- [i.4] ITU-T Recommendation F.400/X.400 (1999): "Message handling system and service overview".

# 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 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:

Address Complete Message **ACM ANM** Answer Message AS **Application Specific** ATC Abstract Test Case Abstract Test Method ATM ATP Access Transport Parameter **Abstract Test Suite** ATS **Backward Call Indicators** BCI

CCBS Completion of Communication to Busy Subscriber

CGB Circuit Group Blocking
CON Connect Message
CPG Call Progress Message
CPS Calling Party's Category
DSS1 Digital Subscriber System No. 1

FCI Forward Call Indicators
GRS Group Reset message
HLC High Layer Compatibility
IAM Initial Address Message

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

O-MGCF Outgoing Media Gateway Control Function
PICS Protocol Implementation Conformance Statement
PIXIT Protocol Implementation eXtra Information for Testing

REL Release Message
RSC Reset Circuit message
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 of 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)	112xxx
	Receipt of the RESUME Message (RES)	113xxx

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 [2].

#### 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)

TP101001	SIP reference: RFC 3261	[6]		SUP reference:
			ES 283 (	027 [1], clause 7.2.3.1.1
TSS reference:	SIP-ISUP/Basic call/ Sending of t	he Initial Add	lress message	(IAM)/
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	Normal call setup without precond	dition require	ment	
	Ensure that if the SIP precondition			
	header, the I-MGCF shall send ar			
	I-MGCF shall set the continuity in	dicators to "C	Continuity chec	k not required".
SIP Parameter				
values:				
ISUP Parameter				
values:				
Comments:	SIP	SL		ISUP
	INVITE -		<b>→</b>	IAM
	180 Ringing ←		<b>←</b>	ACM
		Ringin	g tone	
	200 OK INVITE ←		<b>←</b>	ANM
	ACK →			
		Conver	sation	
	BYE →		<b>→</b>	REL
	200 OK BYE ←		+	RLC

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 Add	lress message (IAM)/		
SIP selection	PICS 4/4 AND PICS 4/5			
criteria:				
ISUP selection				
criteria:	0-11	(		
Test purpose:	Call setup with precondition tag in the Supported header and preconditions are fulfilled 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 fulfilled 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	INVITE: Supported: 100rel, precondition			
values:	SDP a=curr:qos local sendrecv			
	a=curr:qos remote none			
	a=des:qos mandatory local sene a=des:qos none remote sendre			
	a=des.qos none remote sendre	υV		
	200 OK INVITE			
	SDP a=curr:qos local sendrecv			
	a=curr:qos remote sendrecv			
	a=des:qos mandatory local sendrecv			
10115 5	a=des:qos mandatory remote se			
ISUP Parameter values:	IAM: Continuity indicator: Continuity check not	•		
Comments:	SIP SU			
	INVITE -	→ IAM		
	180 Ringing ←	<b>←</b> ACM		
	Ringing			
	200 OK INVITE ←	<b>←</b> ANM		
	7.01	action		
	Conver BYE →	sation → REL		
	200 OK BYE	← RLC		
	ZOU ON DIE	▼ NLO		

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 Ad	dress message (IAM)/			
SIP selection criteria:	NOT PICS 4/4 AND NOT PICS 4/5				
ISUP selection criteria:					
Test purpose:	Call setup with precondition tag in the Supported header and preconditions are fulfilled 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				
CID Donomotor	the precondition requirement.				
SIP Parameter values:	INVITE: Supported: 100rel, precondition SDP a=curr:gos local sendrecv				
values.					
	a=curr:qos remote none a=des:qos mandatory local sendrecv				
	a=des:qos none remote sendre				
ISUP Parameter values:	IAM: Continuity indicator: Continuity check no				
Comments:	SIP SI	JT <b>ISUP</b>			
	INVITE →	→ IAM			
	180 Ringing ←	← ACM			
	Ringing tone				
	200 OK INVITE ←	<b>←</b> ANM			
	ACK →				
	Conve	rsation			
	BYE →	→ REL			
	200 OK BYE <b>←</b>	<b>←</b> 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 A	ddress message (IAM)/		
SIP selection	PICS 4/4 AND PICS 4/5			
criteria:				
ISUP selection	PICS 4/1			
criteria:	Only and the supposed the suppo			
Test purpose:	Call setup with precondition tag in the Require header and requirement for resource reservation			
		econdition tag in the Require header the not fulfilled the I-MGCF shall set the continuity a previous circuit" or "required on this circuit".		
SIP Parameter	INVITE: Require: precondition	a provided direction required on the direction.		
values:	SDP a=curr:gos local none			
	a=curr:qos remote none			
	a=des:qos mandatory local se			
	a=des:qos none remote send	ecv		
	183: Require: 100rel			
	SDP a=curr:qos local none			
	a=curr:qos remote none			
	a=des:gos mandatory local se	endrecv		
	a=des:qos mandatory remote			
	a=conf:qos remote sendrecv			
	UPDATE:			
	SDP a=curr:qos local sendrecv			
	a=curr:qos local sentifect			
	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 se	ndrecv		
	a=des:qos mandatory remote	sendrecv		
ISUP Parameter	IAM: "continuity check required on this circu	t" or "Continuity check performed on a		
values:	previous circuit"	NUT. IOUR		
Comments:		SUT ISUP		
	INVITE  183 Session Progress  ←	→ IAM		
	183 Session Progress ← PRACK →			
	200 OK PRACK			
	200 OK FRACK			
	UPDATE → COT			
	200 OK UPDATE			
	180 Ringing ←	← ACM		
	PRACK →			
	200 OK PRACK ←			
	-	ng tone		
	200 OK INVITE	<b>←</b> ANM		
	ACK →	orgation		
	BYE ->	ersation → REL		
	200 OK BYE ←	→ REL ← RLC		
	ZOU ON DIE	▼ NLO		

TP101005	SIP reference: RFC 3261	[6]		SUP reference: 027 [1], clause 7.2.3.1.1	
TSS reference:	SIP-ISUP/Basic call/ Sending of the	ne Initial Add	ress message	(IAM)/	
SIP selection	PICS 4/4 AND PICS 4/5				
criteria:					
ISUP selection	PICS 4/1				
criteria:					
Test purpose:	Call setup with precondition tag in the Supported header and requirement for resource reservation				
	Ensure if the INVITE request cont received SDP indicates that preceindicators to "continuity check per	ondition is no	ot fulfilled the I-N	MGCF shall set the continuity	
SIP Parameter	INVITE: Supported: precondition				
values:	SDP a=curr:gos local no	ne			
	a=curr:qos remote	none			
	a=des:qos mandato	ory local sen	drecv		
	a=des:qos none rer	note sendre	cv		
	183: Require: 100rel				
	SDP a=curr:qos local no				
	a=curr:qos remote		dan a		
	a=des:qos mandato a=des:qos mandato				
	a=des.qos mandato a=conf:gos remote		endrecv		
	a=con.qos remote	Sendrecv			
	UPDATE:				
	SDP a=curr:qos local sei	ndrecv			
	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 sei				
	a=curr:qos remote				
	a=des:qos mandato				
ISUP Parameter	a=des:qos mandato			ala a ala manda mana al a mana	
values:	IAM: "continuity check required or previous circuit"	i triis circuit	or Continuity	check performed on a	
Comments:	SIP	Sl	IT	ISUP	
Comments.	INVITE →	30	, , , , , , , , , , , , , , , , , , ,	IAM	
	183 Session Progress		•	IAIVI	
	PRACK				
	200 OK PRACK				
	200 ON PRACK				
	UPDATE → COT				
	200 OK UPDATE		•	001	
	200 OK OF DATE				
	180 Ringing ←		<b>←</b>	ACM	
	PRACK		•	, (Olvi	
	200 OK PRACK				
	200 01(110(0))	Ringin	n tone		
	200 OK INVITE ←	i tirigiri	<b>←</b>	ANM	
	ACK →		•	, u vivi	
		Conve	sation		
	BYE →	CONVE	<b>→</b>	REL	
	200 OK BYE		ŕ	RLC	
	ZOO ON DIL			INLO	

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 Add	dress message (IAM)/			
SIP selection	NOT PICS 4/4 AND PICS 4/5				
criteria:					
ISUP selection	PICS 4/1				
criteria:					
Test purpose:	Call setup with precondition tag in the Require header and requirement for resource 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	INVITE: Require: precondition	•			
values:	SDP a=curr:qos local none				
	a=curr:qos remote none				
	a=des:qos mandatory local sendrecv				
	a=des:qos none remote sendre	CV			
ISUP Parameter					
values:					
Comments:	SIP SU	JT <b>ISUP</b>			
	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 Ad					
SIP selection	PICS 4/4 AND PICS 4/5	3 ( )				
criteria:						
ISUP selection	PICS 1/3 AND NOT PICS 4/1					
criteria:						
Test purpose:	COT procedure not supported, IAM delayed	until preconditions met				
	Ensure if Continuity Check procedure is not s					
	in the received INVITE request contains pred					
		are met and set the continuity indicators in the				
	resulting IAM to "Continuity check not require	ed".				
SIP Parameter	INVITE: Require: precondition					
values:	SDP a=curr:qos local none					
	a=curr:qos remote none	ndro ov				
	a=des:qos mandatory local ser a=des:qos none remote sendre					
	a=des.qos none remote sendit	ecv				
	183: Require: 100rel					
	SDP a=curr:gos local none					
	a=curr:gos remote none					
	a=des:qos mandatory local sei	ndrecv				
	a=des:qos mandatory remote s					
	a=conf:qos remote sendrecv					
	UPDATE:					
	SDP a=curr:qos local sendrecv a=curr:qos remote none					
	a=curr.qos remote none a=des:qos mandatory local sendrecv					
	a=des:qos mandatory remotes					
	a accided manager, remote contained.					
	200 OK UPDATE					
	SDP a=curr:qos local sendrecv					
	a=curr:qos remote sendrecv					
	a=des:qos mandatory local ser					
	a=des:qos mandatory remote s					
ISUP Parameter		required on this circuit,				
values:	COT Continuity Indicator: continuity check					
Comments:		UT <b>ISUP</b>				
	INVITE -					
	183 Session Progress ←					
	PRACK -					
	200 OK PRACK ←					
	UPDATE →	→ IAM				
	200 OK UPDATE ←	<b>4</b> ACM				
	180 Ringing ← PRACK →	<b>←</b> ACM				
		ng tono				
	~	ng tone ← ANM				
	200 OK INVITE ← ACK →	AINIVI				
		ersation				
	BYE →	Preservation → 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 Ac	ddress 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 (I	PIXIT)			
ISUP Parameter values:					
Comments:	SIP INVITE  488 Not Acceptable Here  ACK  S  S  ACK  S	SUT <b>ISUP</b>			

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 cor	ntained in a single INVITE request, the				
	I-MGCF shall select one of the supported me					
	media stream, and reject the other media stre					
	answer, as detailed in RFC 3264 [18]. If supp					
	non-audio media stream(s) are contained in a should be selected.	a single INVITE request, an audio stream				
SIP Parameter	Offer: m=audio 4711 RTP/AVP 8					
values:	m= video 4713 RTP/AVP 31					
, alacoi	111- 11000 17 10 1111 7711 01					
	Answer: m=audio 4711 RTP/AVP 8					
	m=video 0 RTP/AVP 31					
ISUP Parameter						
values:						
Comments:		UT <b>ISUP</b>				
	INVITE →	→ IAM				
	180 Ringing ←	← ACM				
		ng tone				
	200 OK INVITE	<b>←</b> ANM				
	ACK →					
		ersation				
	BYE -	→ REL				
	200 OK BYE ←	← RLC				

TP101010	SIP reference: RFC	SIP reference: RFC 3261 [6] ISUP reference: ES 283 027 [1], clause 7 RFC 3264 [18]						
TSS reference:	SIP-ISUP/Basic call/ Sendi	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/						
SIP selection	PICS 4/15							
criteria:								
ISUP selection criteria:								
Test purpose:	To tag included in 183 prov	risional re	sponse					
	response, in order to estab				ckward non-100 provisional n RFC 3261 [6]			
SIP Parameter	183 To taq included							
values:								
ISUP Parameter	ACM: oBCi "inband info ava	ailable"						
values:								
Comments:	SIP		SUT		ISUP			
	INVITE	<b>→</b>		→	IAM			
	183 Session Progress	<b>←</b>		<b>←</b>	ACM(no indication)			
	180 Ringing	<b>←</b>		<b>←</b>	CPG(Alerting)			
			Ringing tone					
	200 OK INVITE	<b>←</b>		<b>←</b>	ANM			
	ACK	<b>→</b>						
			Conversation					
	BYE	<b>→</b>		<b>→</b>	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 A	ddress message (IAM)/
SIP selection		<u>-</u>
criteria:		
ISUP selection		
criteria:		
Test purpose:	To tag included in 180 provisional response	
		tag in the first backward non-100 provisional
	response, in order to establish an early dialog	og as described in RFC 3261 [6]
SIP Parameter	180 To tag included	
values:		
ISUP Parameter		
values:		
Comments:	SIP	SUT ISUP
	INVITE →	→ IAM
	180 Ringing ←	<b>←</b> ACM
	Ring	ing tone
	200 OK INVITE ←	← ANM
	ACK →	
	Conv	ersation
	BYE →	→ REL
	200 OK BYE ←	← RLC

TP101012	SIP reference: RFC 3261 [6]		eference:			
			auses 7.2.3.1.2.2 and 3.1.2.3			
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/					
SIP selection	PICS 2/1					
criteria:						
ISUP selection	PICS NOT 4/16					
criteria:		1 11: 1: 4				
Test purpose:	Setting of nature of connection indicator and to	orward call indicator				
	Ensure that the SUT on receipt of an INVITE	nessage:				
	sends an IAM message, where the Calling pa subscriber", the <b>Nature of Connection Indic</b>	ntors (NCI) encoded a	as follows:			
	<ul> <li>Satellite indicator set to: One satellite circ</li> </ul>					
	Echo control device indicator set to: "Out		vice included".			
	The Forward call indicator is encoded a					
	Interworking indicator: Interworking encountries ISUP/BICC Indicator: ISDN User part/BIC		N/			
	ISUP/BICC Preference indicator: ISDN u					
	ISDN access indicator: Originating acces		anod an tho way.			
SIP Parameter						
values:						
ISUP Parameter	Nature of Connection Indicators (NCI):					
values:	Satellite indicator set to: "One satellite circ					
	Echo control device indicator set to: "Outgoing echo control device included"					
	Forward Call Indicators (FCI):					
	Interworking indicator: interworking encountered					
	ISDN user part indicator: ISDN user part/E		way			
	ISDN access indicator: originating access		•			
	ISDN user part preference indicator: ISDN					
Comments:	SIP SU					
	INVITE →	→ IAM				
	180 Ringing ←	← ACN	<b>VI</b>			
	Ringin					
	200 OK INVITE ←	<b>←</b> ANN	VI			
	ACK → Conve	cation				
	BYE →	salion → REL				
	200 OK BYE	← RLC				
	ZOU ON DIL	₹ NLC	,			

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 Add	dress message (IAM)/			
SIP selection	PICS 1/1				
criteria:					
ISUP selection					
criteria:					
Test purpose:	Setting of nature of connection indicator and f	orward call indicator T38 codec received			
	Ensure that the SUT on receipt of an INVITE	•			
	sends an IAM message, where the Calling pa subscriber", the <b>Nature of Connection Indica</b>	ators (NCI) encoded as follows:			
	Satellite indicator set to: "One satellit				
	<ul> <li>Echo control device indicator set to: '         <ul> <li>the Forward call indicator is encode</li> </ul> </li> </ul>	"Outgoing echo control device <b>not</b> included".			
	Interworking indicator: Interworking e				
	ISUP/BICC Indicator: ISDN User par				
		ON user part/BICC not required all the way			
	ISDN access indicator: Originating a				
SIP Parameter	INVITE with SDP m line T:38				
values:					
ISUP Parameter	Nature of Connection Indicators (NCI):				
values:	Satellite indicator set to: "One satellite circ	uit in the connection"			
	Echo control device indicator set to: "Outgoing echo control device <b>not</b> included"				
	Forward Call Indicators (FCI):				
	Interworking indicator: interworking encountered				
	ISDN user part indicator: ISDN user part/B				
	ISDN access indicator: originating access				
	ISDN user part preference indicator: ISDN				
Comments:	SIP SU				
	INVITE -	→ IAM			
	180 Ringing ←	← ACM			
	Ringin				
	200 OK INVITE ← ACK →	<b>←</b> ANM			
	Conve	raction			
	BYE → Conver	REL → REL			
	200 OK BYE	→ REL ← RLC			
	ZUU UN DIE <b>T</b>	₹ KLU			

TP101014	SIP reference: RFC 3261 [6]		UP reference: 7 [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	PICS 2/3					
criteria:						
ISUP selection	PICS 4/16					
criteria: Test purpose:	Setting of nature of connection indicator and	orward call indice	ator indicating ISDN and			
rest purpose.	TMR 64 kBit/s		-			
	Ensure that the SUT on receipt of an INVITE	message with SD	PP m line CLEARMODE:			
	<ul> <li>sends an IAM message, where the Callin subscriber", if the TMR = 64 kBit/s unrest Indicators (NCI) encoded as follows:</li> </ul>					
	- the Nature of Connection Indicator	rs (NCI) encoded	l as follows:			
	<ul> <li>Satellite indicator set to: "One satelli</li> </ul>					
	- Echo control device indicator set to:		entrol device not included.			
	the Forward call indicator is encod     Interworking indicator: No interworking					
	ISUP/BICC Indicator: ISDN User par		ne way			
	ISUP/BICC Preference indicator: ISI					
	ISDN access indicator: Originating access ISDN.					
SIP Parameter						
values:						
ISUP Parameter	Nature of Connection Indicators (NCI):					
values:	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 encountered					
	ISDN user part indicator: ISDN user part/E		way			
	ISDN access indicator: originating access ISDN					
Comments:	ISDN user part preference indicator: ISDN SIP		ISUP			
Comments:	SIP SUNVITE →	) i →	IAM			
	180 Ringing	<del>-</del>	ACM			
	Ringin	<del>-</del>	AOW			
	200 OK INVITE	<del>←</del>	ANM			
	ACK →					
	Conve	rsation				
	BYE →	<b>→</b>	REL			
	200 OK BYE <b>←</b>	+	RLC			

TP101015	SIP reference: RF	FC 3261 [6]	-	SUP reference: 27 [1], clause 7.2.3.1.2.5		
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/					
SIP selection criteria:	Based on table 1					
ISUP selection criteria:						
Test purpose:	Mapping of SDP into the Ensure that the SUT in the media description defined a_b_m_LINE_VALUE: sends an IAM message, v to TMR_VALUE.	e Idle state on receip in table 1 with the "a	a =" "b =" and "			
SIP Parameter values:	INVITE; a_b_m_LINE_VA	LUE				
ISUP Parameter values:	IAM; TMR: ISUP_TMR					
Comments:	SIP	SU	JT	ISUP		
	INVITE	<b>→</b>	<b>→</b>	IAM		
	180 Ringing	<b>←</b>	<b>←</b>	ACM		
		Ringin	g tone			
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM		
	ACK	<b>→</b>				
		Conve	sation			
	BYE	<b>→</b>	<b>→</b>	REL		
	200 OK BYE	+	+	RLC		

Table 1

			Value	es for test purposes TP1	01015	
			a_b_m	_LINE_VALUE		
		m= line		b= line	a= line	TMR_VALUE
test purposes	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwidt h-value=""></bandwidt></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters&gt;</clock></encoding></dynamic-pt>	TMR codes
VA_01	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> PCMU/8000</dynamic-pt>	"3,1 KHz 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</dynamic-pt>	"3,1 KHz audio"
VA_04	audio	RTP/AVP	Dynamic PT	AS: 64 kbit/s	rtpmap: <dynamic-pt> CLEARMODE/8000 (NOTE 2)</dynamic-pt>	"64 kbit/s unrestricted"
VA_05	image	udptl	t38 [31]	N/A or up to 64 kbit/s	Based on ITU-T Recommendation T.3 8 [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.3 8 [31]	"3,1 kHz audio"
VA_07	audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	"3,1 KHz 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.

SIP ref	ference: RFC 3261 [6]	ES 283	ISUP reference: 027 [1], clause 7.2.3.1.2.5		
SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/					
PICS 2/3	<u> </u>				
PICS 4/18 AN	D PICS 4/19				
elements into  Ensure that whelement:  If the first state as stated within the I-MGCF shand shall set the first BC:  second BC:  USI Prime:	TMR and USI prime  then the INVITE request included codec in the INVITE is a in the second Bearer Capable and the Second Bearer Capable TMR to "64 kBit/s prefer 3,1 kHz audio or speech unrestricted digital information unrestricted digital information."	udes multiple PS codec appearing ility in the XML B pability element ed". on with tones/an	TN XML bearer information in table 1 and is the equivalent searer Capability element then into the TMR and USI prime		
	64 kBit/s preferred	01.17	LOUID		
			ISUP		
	<del>-</del>	-	, ,,,,,,		
180 Ringing	<del>-</del>	•	E ACM		
200 OK INIVIT		• •	- ANM		
	<del>-</del>	•	AINIVI		
ACK	=	overeation			
RYE			REL		
	÷	-	RLC		
	SIP-ISUP/Bas PICS 2/3  PICS 4/18 AN  Fallback connelements into  Ensure that wellement:  If the first state as stated within the I-MGCF stand shall set to first BC: second BC: USI Prime: TMR: SIP INVITE 180 Ringing	PICS 2/3  PICS 4/18 AND PICS 4/19  Fallback connection type supported: Mappelements into TMR and USI prime  Ensure that when the INVITE request include element:  If the first stated codec in the INVITE is a castated within the second Bearer Capabethe I-MGCF shall map the XML Bearer	SIP-ISUP/Basic call/ Sending of the Initial Address message PICS 2/3  PICS 4/18 AND PICS 4/19  Fallback connection type supported: Mapping of the second elements into TMR and USI prime  Ensure that when the INVITE request includes multiple PS element:  If the first stated codec in the INVITE is a codec appearing as stated within the second Bearer Capability in the XML Bearer Capability element and shall set the TMR to "64 kBit/s preferred".  first BC: 3,1 kHz audio or speech second BC: unrestricted digital information with tones/an TMR: 64 kBit/s preferred  SIP SUT INVITE SUT		

TP101017	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 Add	dress message (IAM)/	
SIP selection	PICS 4/18	<u>-</u>	
criteria:			
ISUP selection	PICS 4/19		
criteria:			
Test purpose:	Fallback connection type supported: Mapping elements into USI and TMR prime	·	
	Ensure when the INVITE request includes mu MGCF shall:	Itiple PSTN XML BearerCapability then the I-	
	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 Ca		
	then the I-MGCF shall map the XML Bearer C		
	USI and shall map the TMR prime from the PS (InformationTransferCabability).	STN XML BearerCapability	
SIP Parameter	first BC: 3,1 kHz audio or speech		
values:	second BC: unrestricted digital information w	ith tones/announcements	
ISUP Parameter	USI: 3,1 kHz audio or speech		
values:	TMR prime: 3,1 kHz audio or speech		
Comments:	SIP SU		
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	Ringin	<u> </u>	
	200 OK INVITE ←	← ANM	
	ACK →		
	Conve		
	BYE →	→ REL	
	200 OK BYE <b>←</b>	<b>←</b> RLC	

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 Add	lress message (IAM)/	
SIP selection	PICS 4/18		
criteria:			
ISUP selection	PICS 4/19		
criteria:			
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 mu MGCF shall:	Itiple PSTN XML BearerCapability then the I-	
	if the compared first codec stated within the IN		
	second XML Bearer Capability element, then	the second XML Bearer Capability element	
OID D	shall be discarded.		
SIP Parameter values:	PSTN XML BC 1 (speech or 3,1 kHz audio)	ion with tanca and announcements)	
values.	PSTN XML BC 2 (unrestricted digital information with tones and announcements)		
	SDP: m =audio xxx, RTP/AVP 0 8		
ISUP Parameter	USI: not included		
values:	TMR. 3,1 kHz audio		
Comments:	SIP SU	IT <b>ISUP</b>	
	INVITE →	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
	Ringin		
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conver		
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

TP101019	SIP reference: RFC 3261 [6]	ı	SUP reference:
		ES 283 0	27 [1], clause 7.2.3.1.2.5
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial	Address message	(IAM)/
SIP selection	PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of PSTN XML BearerCapability 6	lement into the US	I parameter
	Ensure that the SUT in the Idle state on re		
	BearerCapability element the mapping of t	he USI ISUP_USI	shall be taken from the PSTN
	XML BearerCapability value ISDN_BC.		
SIP Parameter	PSTN XML BearerCapability ISDN_BC		
values:			
ISUP Parameter	IAM: USI = ISUP_USI		
values:			
Comments:	SIP	SUT	ISUP
	INVITE →	<b>→</b>	IAM
	180 Ringing ←	<b>←</b>	ACM
	Rir	ging tone	
	200 OK INVITE ←	<b>←</b>	ANM
	ACK →	_	
		versation	
	BYE →	→	REL
	200 OK BYE ←	<b>+</b>	RLC
	ZUU UN DIE		KLU

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

TP101020	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 A	Address message (IAM)/	
SIP selection	PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of PSTN XML BearerCapability el	lement into the <b>TMR</b> parameter	
	PSTN XML BearerCapability value ISDN_E sends an IAM message, with the Transmiss to ISUP_TMR.	ceipt of an INVITE message containing one BC_ITC sion Medium Requirement (TMR) parameter set	
SIP Parameter	INVITE; PSTN XML BearerCapability		
values:			
ISUP Parameter values:	IAM: TMR		
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
	Ring	ging tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Con	versation	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

Values and selection criteria for the test purpose TP101020			
VA_01	ISDN_BC_ITC = speech	ISUP_TMR = speech	
	ISDN_BC_ITR = 64 kbits/s		
VA_02	ISDN_BC_ITC = 3,1 kHz audio	ISUP_TMR = 3,1 kHz audio	
	ISDN_BC_ITR = 64 kbits/s		
VA_03	ISDN_BC_ITC = unrestricted digital information	ISUP_TMR = 64 kbits/s unrestricted	
	ISDN_BC_ITR = 64 kbits/s		

TP101021	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 Ad	ddress message (IAM)/
SIP selection	PICS 4/18	
criteria:		
ISUP selection criteria:		
Test purpose:	No PSTN XML received, mapping of HLC in	ATP
	<ul> <li>Ensure that the SUT in the Idle state on recedescription defined in table 2 with the "a =" "a_b_m_LINE_VALUE:</li> <li>sends an IAM message with the Access information element.</li> </ul>	
SIP Parameter values:	INVITE: a_b_m_LINE_VALUE	
ISUP Parameter values:	IAM; Access transport parameter HLC: HL	C_VALUE
Comments:	SIP	SUT ISUP
Comments.	INVITE →	→ IAM
	·····	← ACM
	100111191119	1
	•	ng tone
	200 OK INVITE ←	<b>←</b> ANM
	ACK →	
	Conv	ersation
	BYE →	→ REL
	200 OK BYE ←	← RLC

Table 2

•	•		Valu	es for test purposes TP10102	21	
M= line				b= line	a= line	HLC parameter HLC VALUE
Test purposes	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwidth-value></bandwidth-value></modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters&gt;</clock></encoding></dynamic-pt>	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</dynamic-pt>	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</dynamic-pt>	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	
	OUT:		
	Ensure that the SUT in the Idle state on receip BearerCapability BC_VALUE and the PSTN >		
	bearerCapability BC_VALUE and the PSTN 7	TWIL HIGHLayerCompatibility HLC_VALUE	
	sends an IAM message with the Access tran	sport parameter containing the received	
	PSTN XML HighLayerCompatibility].	oport parameter containing the received	
SIP Parameter	INVITE		
values:	PSTN XML BearerCapability: BC_VALUE		
	PSTN XML HighLayerCompatibility: HLC_VALUE		
ISUP Parameter	IAM; Access transport parameter HLC: HLC_VALUE		
values:			
Comments:	SIP St		
	INVITE →	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
	Ringin	•	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conve		
	BYE →	→ REL	
	200 OK BYE <b>←</b>	<b>←</b> 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 [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
	[i.2])
	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 [i.1])
	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
	[i.4])
	BC_VALUE = Unrestricted digital information
VA_11	HLC_VALUE = OSI application (X.200 - Series ITU-T Recommendations [i.3])
	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 Ad	ldress message (IAM)/
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	sends an IAM message with the User Telese	ipt of a INVITE message with PSTN XML  I XML HighLayerCompatibility HLC_VALUE
	PSTN XML HighLayerCompatibility.	
SIP Parameter	INVITE	
values:	PSTN XML BearerCapability: BC_VALUE PSTN XML HighLayerCompatibility: HLC_	VALUE
ISUP Parameter values:	IAM; User teleservice parameter	
Comments:	SIP S	UT <b>ISUP</b>
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Ringir	ng tone
	200 OK INVITE ←	<b>←</b> ANM
	ACK →	
		ersation
	BYE →	→ REL
	200 OK BYE ←	<b>←</b> 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
	[i.2])
	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 [i.1])
	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
	[i.4])
	BC_VALUE = Unrestricted digital information
VA_11	HLC_VALUE = OSI application (X.200 - Series Recommendations [i.3])
	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]	=	SUP reference:
TCC reference:	ES 283 027 [1], clause 7.2.3.1.2.10		
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial A	adress message	(IAM)/
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of PSTN XML LowLayerCompatib	ility in ATP LLC	
	Ensure that the SUT in the Idle state on rec LowLayerCompatibility sends an IAM message with the Access tra LowLayerCompatibility as received in the	ansport paramete INVITE message	er containing the PSTN XML
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 →	<b>→</b>	IAM
	180 Ringing ←	<b>←</b>	ACM
	0 0	ing tone	-
	200 OK INVITE ←	<b>+</b>	ANM
	ACK →	•	7 (1 41)
		versation	
	BYE →	-	REL
	I	-	· ·
	200 OK BYE ←	<b>←</b>	RLC

TP101025	SIP reference: R	FC 3261 [6]	I	SUP reference:
			ES 283 02	27 [1], clause 7.2.3.1.2.10
TSS reference:	SIP-ISUP/Basic call/ Sen	ding of the Initial Add	dress message	(IAM)/
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	Mapping of PSTN XML P	ProgressIndicator in A	NTP PI	
		e Idle state on receip	ot of a INVITE r	nessage with a valid <b>PSTN</b>
	XML ProgressIndicator			
	sends an IAM message with the Access transport parameter containing the PSTN XML			
	ProgressIndicator as re-			
SIP Parameter	INVITE; PSTN XML ProgressIndicator: PI_VALUE			
values:				
ISUP Parameter	IAM; progress indicator PI_VALUE			
values:				
Comments:	SIP	SL	JT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM
		Ringin	g tone	
	200 OK INVITE	<b>←</b>	<b>+</b>	ANM
	ACK	<b>→</b>		
		Conver	rsation	
	BYF	<b>→</b>	<b>→</b>	REL
	200 OK BYE	<del>-</del>	<b>É</b>	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)		

TP101026	SIP reference: F	RFC 3261 [6]		ISUP reference: 127 [1], clause 7.2.3.1.2.9
TSS reference:	SIP-ISUP/Basic call/ Se	nding of the Initial Add	dress message	(IAM)/
SIP selection criteria:				
ISUP selection criteria:	PICS 4/3			
Test purpose:	HOP counter derived from the Max-Forward header  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.			
SIP Parameter	Max-Forward header			
values:				
ISUP Parameter values:	IAM: Hop Counter para	ameter value		
Comments:	SIP	SUT		ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing Ringing tone	<b>←</b>	<b>←</b>	ACM
	200 OK INVITE ACK Conversation	<b>←</b> →	<b>←</b>	ANM
	BYE	<b>→</b>	<b>→</b>	REL
	200 OK BYE	<del>-</del>	<del>-</del>	RLC
				should be large enough to expected of a validly routed

TP101027	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 Ad	ldress message (IAM)/		
SIP selection				
criteria:				
ISUP selection	NOT PICS 1/5			
criteria:				
Test purpose:	Mapping of Request URI into called party nu	mber "national number"		
	Analyse the information contained in receive			
		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.			
SIP Parameter	INVITE: Request URI			
values:				
ISUP Parameter	IAM: Called party number			
values:				
Comments:		UT ISUP		
	INVITE →	→ IAM		
	180 Ringing ←	← ACM		
	Ringi	ng tone		
	200 OK INVITE ←	<b>←</b> ANM		
	ACK →			
	Conv	ersation		
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

TP101028	SIP reference: RFC	3261 [6]	_	SUP reference:
			ES 283 02	27 [1], clause 7.2.3.1.2.1
TSS reference:	SIP-ISUP/Basic call/ Sendin	g of the Initial Add	lress message	(IAM)/
SIP selection				
criteria:				
ISUP selection	PICS 1/5			
criteria:				
Test purpose:	Mapping of Request URI into	o called party nun	nber "internatioi	nal number"
	Analyse the information cont			
	If CC is not the country code			hop terminates, then set
	Nature of Address indicator	to "International	number".	
SIP Parameter	INVITE: Request URI			
values:				
ISUP Parameter	IAM: Called party number			
values:	0.0		_	
Comments:	SIP	SL		ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM
		Ringin	•	
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
		Conver	sation	
	BYE	<b>→</b>	<b>→</b>	REL
	200 OK BYE	<b>←</b>	<b>←</b>	RLC

TP101029	SIP reference: RFC 3261 [6]	ISUP reference:
TCC reference:	CID ICLID/Dania call/ Conding of the Initial Ada	ES 283 027 [1], clause 7.2.3.1.2.1
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Add	ress message (IAM)/
SIP selection		
criteria:		
ISUP selection		
criteria:		
Test purpose:	Mapping of Request URI into called party numnetwork number indicator  Ensure that the SUT on receipt of an INVITE recontained in the userinfo component of the Re  Internal Network Number Indicator: routine Numbering plan Indicator: 001 ISDN (Tele	message with a Called party number equest-URI: g to internal network number not allowed.
values:	INVITE. Request ORI	
ISUP Parameter values:	IAM: Called party number	
Comments:	SIP SU	IT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Ringing	g tone
	200 OK INVITE ←	<b>←</b> ANM
	ACK →	
	Conver	rsation
	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:	number CC is the same as the MGCF is loan.  Analyse the information contained in rece	ved E.164 address. ich the next hop terminates, then remove "+CC"
SIP Parameter	INVITE: Request URI	
values:	'	
ISUP Parameter values:	IAM: Called party number address signa	ls
Comments:	SIP	SUT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Rii	nging tone
	200 OK INVITE ←	<b>←</b> ANM
	ACK →	
	Со	nversation
	BYE →	→ REL
	200 OK BYE ←	<b>←</b> 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 Request Unumber CC is not the same as the MGCF in Analyse the information contained in received to the country code of the network.	is located	
	"+" and use the remaining digits to fill the Address signals.		
SIP Parameter	INVITE: Request URI	<u> </u>	
values:	·		
ISUP Parameter values:	IAM: Called party number address signal	S	
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	Rin	ging tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Con	versation	
	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 Add	dress message (IAM)/	
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of calling party category		
	Ensure that a cpc SIP_CPC parameter SIP_C		
	parameter and the "language" in the Accept-L		
	the calling partycategory parameter ISUP_CPC in the sent IAM		
SIP Parameter	INVITE: P-Asserted-Identity = PARAM, Accept-Language = ISUP_CPC		
values:			
ISUP Parameter	IAM: Calling Party Category		
values:			
Comments:	SIP	JT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	Ringin	g tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conve	rsation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

Values for test purposes TP101032			
SIP_CPC		ISUP_CPC	
cpc received in a P-Asserted-Identity PARAM	Accept-Language SIP_LANG	Sent Calling party's category	
operator	fr	operator, language French	
operator	en	operator, language English	
operator	de	operator, language German	
operator	ru	operator, language Russian	
operator	es	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	

TP101033	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 Add	lress message (IAM)/	
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of PSTN XML ProgressIndicator #6 into FCi ISDN		
	Ensure that when a PSTN XML ProgressIndicator value #6 is received in a INVITE MIME		
	body, that the forward call indicator in the sent IAM idicatedes that the originating access is		
	ISDN		
SIP Parameter	INVITE: PSTN XML PI#6		
values:			
ISUP Parameter	IAM: Forward call indicator:		
values:	Interworking indicator: no interwork		
	ISDN User Part indicator: ISDN use		
	ISDN access indicator: originating a	access ISDN	

TP101033	SIP reference:	RFC 3261 [6]		ISUP reference: 27 [1], clause 7.2.3.1.2.1
Comments:	SIP	;	SUT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM
		Ring	ing tone	
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
		Conv	ersation/	
	BYE	<b>→</b>	<b>→</b>	REL
	200 OK BYE	<b>←</b>	<b>←</b>	RLC

## 6.2.1.2 Overlap procedure at the I-MGCF

TP102001	SIP reference: RFC 32	61 [6]		Į.	SUP reference:
					27 [1], clause 7.2.3.1.2.1
TSS reference:	SIP-ISUP/Basic call/Sending of the Subsequent address message (SAM)/				
SIP selection					
criteria:					
ISUP selection					
criteria:			2005		
Test purpose:	Overlap sending successful at a	the I-M	GCF		
	Ensure that the SUT receives a	ا/\اها ما	TE with the som	o Coll I	D and From tag as a provious
	INVITE which was associated v				
	existing on the BICC/ISUP side				
	greater than the number of digi				
			,		
	<ul> <li>sends a SAM and pass it to</li> </ul>	o outgo	ing BICC/ISUP	proced	ures.The SAM shall contain in
	its Subsequent Number pa				
	Request-URI compared wi	th the	digits already ac	cumula	ted for the call.
SIP Parameter					
values:					
ISUP Parameter					
values:					
Comments:	SIP	_	SUT	_	ISUP
	INVITE(CS1)	<b>→</b>		<b>→</b>	IAM
	IN IV (ITE (OOO)				0.444
	INVITE(CS2)	<b>→</b>		<b>→</b>	SAM
	484 Address Incomlete(CS1)	<b>←</b>			
	ACK	7			
	INVITE(CS3)	<b>→</b>		<b>→</b>	SAM
	484 Address Incomlete(CS2)	<b>←</b>		_	37 ttv1
	ACK	À			
		-			
	180 Ringing(CS3)	<b>←</b>		<b>←</b>	ACM
		-	Ringing tone	-	
	200 OK INVITE	<b>←</b>		<b>←</b>	ANM
	ACK	<b>→</b>		-	
		-	Conversation		
	BYE	<b>→</b>		<b>→</b>	REL
	200 OK BYE	<del>(</del>		<del>-</del>	RLC

# 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	20 200 027 [1], 010000 712101110	
SIP selection	PICS 4/4 AND PICS 4/5		
criteria:			
ISUP selection	PICS 1/5 AND PICS 4/1		
criteria:			
Test purpose:	COT is sent after precondition met		
	,		
	If the IAM has already been sent, the Continui		
	"continuity check successful", when all of the t		
	- The requested preconditions (if any) in th		
	<ul> <li>A possible outstanding continuity check procedure is successfully performed on the</li> </ul>		
SIP Parameter	outgoing circuit.		
values:	INVITE: Require: precondition SDP a=curr:qos local none		
values.	a=curr:qos remote none		
	a=des:qos mandatory local sen	drecy	
	a=des:qos none remote sendre		
	183: Require: 100rel		
	SDP a=curr:qos local none		
	a=curr:qos remote none		
	a=des:qos mandatory local sen		
	a=des:qos mandatory remote so a=conf:qos remote sendrecv	enarecv	
	a=coni.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:gos local sendrecv		
	a=curr:qos remote sendrecv		
	a=des:qos mandatory local sen	drecv	
	a=des:qos mandatory remote s		
ISUP Parameter	IAM: "Continuity check performed on a previous	us circuit" or "Continuity check required on	
values:	this circuit"		
0	COT continuity indicator: Continuity check suc		
Comments:	SIP SL		
	INVITE  183 Session Progress  ←	→ IAM	
	183 Session Progress ← PRACK →		
	200 OK PRACK		
	UPDATE +	→ COT	
	200 OK UPDATE	<b>9</b> CO1	
	180 Ringing	<b>←</b> ACM	
	PRACK	/ NOIVI	
	200 OK PRACK		
	Ringin	a tone	
	200 OK INVITE	← ANM	
	ACK →		
	Conver	sation	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

# 6.2.1.4 Receipt of the Address Complete Message (ACM)

TP104001	SIP reference: R	FC 3261 [6]	_	SUP reference:
				027 [1], clause 7.2.3.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/			
SIP selection	NOT PICS 4/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	Early ACM not interworke	ed		
	Ensure that the SUT on r		essage where the	ne Called party status
	indicator is set to "no indi	cation":		
	<ul> <li>the ACM is not in</li> </ul>	iterworked.		
SIP Parameter				
values:				
ISUP Parameter	ACM Called party status:	no indication;		
values:				
Comments:	SIP	SU	JT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
			+	ACM (no indication)
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
		Conve	rsation	
	BYE	<b>→</b>	<b>→</b>	REL
	200 OK BYE	<u>+</u>	<b>É</b>	RLC

TP104002	SIP reference: RFC 3261 [6]	ISUP reference:
T00 (		ES 283 027 [1], clause 7.2.3.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address of	complete message (ACM)/
SIP selection	NOT PICS 4/15	
criteria:		
ISUP selection criteria:		
Test purpose:	Early ACM not interworked. Announcement p	rovided by the terminating network
	Ensure that the SUT on receipt of an ACM me indicator is set to "no indication" and during PSTN/ISDN provides an announcement:  • the ACM is not interworked.	
SIP Parameter		
values:		
ISUP Parameter	ACM Called party status: no indication;	
values:		
Comments:		JT ISUP
	INVITE →	→ IAM
		<ul> <li>ACM (no indication)</li> </ul>
	Tones or an	nouncement
	200 OK INVITE ←	← ANM
	ACK →	
	Conve	rsation
	BYE →	→ REL
	200 OK BYE ←	<b>←</b> 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 of	complete message (ACM)/	
SIP selection	PICS 4/15 AND PICS 2/1		
criteria:			
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		
SIP Parameter values:	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.  INVITE: P-Early-Media header, SDP audio xxxx RTP/AVP 8  183 Session Progress: P-Early-Media header		
ISUP Parameter	ACM; CPS indicator: no indication,		
values:	ISUP indicator: ISUP is used all the way		
	ISDN access indicator: ISDN		
	OBCI in-band information: Yes		
Comments:	SIP		
	INVITE -	→ IAM	
	183 Session Progress ←	← ACM (no indication)	
	200 OK INVITE ← ACK →	<b>←</b> ANM	
	Conve		
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> 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	PICS 4/15 AND PICS 2/1		
criteria:			
ISUP selection			
criteria:			
Test purpose:	Early ACM BCI "ISDN User Part not used all t	he way" received. Sending of 183 containing	
	a P-Early-Media header		
	Francisco that OHT are receipt of an AOM receipt	and the opposite direction in a set to the	
	Ensure that SUT on receipt of an ACM messa indication", the <b>ISUP indicator</b> is set to "ISUP		
	and if the I-MGCF has received the P-Early-M		
	not already sent a provisional response includ		
	indicating authorization of early media, then the		
	indicating authorization of ourly modia, then the Fividor Shall.		
	send the 183 Session Progress response with P-Early Media and a		
	P-Early-Media header authorizing early media.		
SIP Parameter	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8		
values:	183 Session Progress: P-Early-Media header		
ISUP Parameter	ACM; CPS indicator: no indication,		
values:	ISUP indicator: ISUP is not used all the way"		
	<b>OBCI</b> with the in-band information: Yes		
Comments:	SIP SU		
	INVITE ->	→ IAM	
	183 Session Progress ←	<ul> <li>ACM (no indication)</li> </ul>	
	200 OK INVITE ←	← ANM	
	ACK →		
	Conver		
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

TP104005	SIP reference: RFC 3261 [6]	-	SUP reference:
<b>700</b> (			27 [1], clause 7.2.3.1.4A
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/		
SIP selection	PICS 2/1 AND PICS 4/18		
criteria:			
ISUP selection criteria:			
	Early ACM BCI "Terminating access non-ISDI	V" rossiyad Ca	anding of 192 containing o
Test purpose:	PSTN XML ProgressIndicator #2	v received. Se	ending of 163 containing a
	Ensure that the SUT, on receipt of an ACM me "no indication", the <b>ISUP indicator</b> is set to "IS"	SUP is used all	
	indicator is set to "non-ISDN", then the I-MG0	CF shall:	
	send the 183 Session Progress respo body containing the progress descript		
SIP Parameter	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8		
values:	183 Session Progress: PSTM XML ProgressIndicator		
ISUP Parameter	ACM: CPS indicator: no indication,		
values:	ISUP indicator: ISUP is used all the way		
	ISDN access indicator: non-ISDN		
	<b>OBCI</b> with the in-band information: No		
Comments:	SIP SUT		ISUP
	INVITE →	<b>→</b>	IAM
	183 Session Progress ←	+	ACM (no indication)
	200 OK INVITE ← ACK →	<b>←</b>	ANM
	Conversation		
	BYE →	<b>→</b>	REL
	200 OK BYE <b>←</b>	<del>-</del>	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 c	omplete message (ACM)/	
SIP selection	PICS 2/1 AND PICS 4/18		
criteria:			
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 <b>CPS indicator</b> is set to "no indication", the <b>ISUP indicator</b> is set to "ISUP not used all the way", then the I-MGCF shall:		
	<ul> <li>send the 183 Session Progress respo progress descriptions "call is not end- information is available in-band (#1).</li> </ul>	o-end ISDN, further call progress	
SIP Parameter	INVITE: P-Early-Media header , SDP audio xxxx RTP/AVP 8		
values:	183 Session Progress: P-Early Media and PSTN XML ProgressIndicator body containing		
	the progress descriptions "call is not end-to-er available in-band (#1)	nd ISDN, further call progress information is	
ISUP Parameter	ACM; CPS indicator: no indication,		
values:	ISUP indicator: ISUP not used all the	wav	
Comments:	SIP SU	,	
	INVITE ->	→ IAM	
	183 Session Progress ←	<ul> <li>ACM (no indication)</li> </ul>	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conver	sation	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> 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 c	omplete 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 <b>CPS indicator</b> is set to "no indication", the <b>OBCI</b> is set to "Inband-info available" then the I-MGCF shall:		
	<ul> <li>send the 183 Session Progress response PSTN XML body containing the progress descriptions "in-band information or an appropriate pattern is now available" (#8).</li> </ul>		
SIP Parameter	INVITE: P-Early-Media header, SDP audio xx		
values:	183 Session Progress: P-Early Media and PS		
	the progress descriptions "in-band information (#8)	or an appropriate pattern is now available"	
ISUP Parameter	ACM; CPS indicator: no indication,		
values:	OBCI: in-band information or an appro	priate pattern is now available	
Comments:	SIP SL	JT ISUP	
	INVITE →	→ IAM	
	183 Session Progress ←	<ul> <li>ACM (no indication)</li> </ul>	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conver	rsation	
	BYE →	→ REL	
	200 OK BYE <b>←</b>	<b>←</b> 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	PICS 2/1 AND PICS 4/18			
criteria:				
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 <b>CPS indicator</b> is set to "no indication", the <b>ISUP indicator</b> is set to "ISUP used all the way", the <b>ISDN access</b>		
	indicator set to PI VALUE:	or raiameter (ATT) containing the progress		
	— — — — — — — — — — — — — — — — — — —	age with PSTN XML ProgressIndicator set		
	to PI_VALUE".	.90		
SIP Parameter	183 Session Progress message PSTN XML I	ProgressIndicator set to PI_VALUE,		
values:				
ISUP Parameter	ACM, CPS indicator: no indication (00)			
values:	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			
Comments:	SIP SI			
	INVITE ->	→ IAM		
	183 Session Progress ←	← ACM (no indication, ATP)		
	200 OK INVITE ←	← ANM		
	ACK →			
	Conve	rsation		
	BYE →	→ REL		
	200 OK BYE <b>←</b>	<b>←</b> RLC		

TP104009	SIP reference: RFC 3261 [6]		ISUP reference: 027 [1], clause 7.2.3.1.4A
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/		
SIP selection	PICS 2/1 AND PICS 4/18	•	
criteria:			
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 <b>interworking indicator</b> I set to no interworking encountered, the <b>ISUP indicator</b> is set to "ISUP is used all the way", the <b>ISDN access indicator</b> is set to "ISDN" and the:		
	send the 183 Session Progress respo progress descriptions "Terminating ac-	cess ISDN"(#	±7).
SIP Parameter values:	183 Session Progress message PSTN XML P	rogressIndic	ator set to value #7
ISUP Parameter	ACM: CPS indicator: no indication		
values:	interworking indicator: no interworking enco	untered	
	ISUP indicator: ISUP used all the way	DDN	
Comments:	ISDN access indicator: terminating access IS	SDN	IOLID
Comments:	SIP SUT INVITE →	<b>→</b>	ISUP IAM
	·····	<del>7</del>	
	183 Session Progress ←	~	ACM (no indication)
	200 OK INVITE	<b>←</b>	ANM
	ACK →		
	Conversation	_	
	BYE -	<b>→</b>	REL
	200 OK BYE <b>←</b>	<del>-</del>	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 of	
SIP selection criteria:	on rear / Busine sam reasonpt of the readings	omplete message (view)
ISUP selection criteria:		
Test purpose:	ACM subscriber free received, a 180 is sent	
	ISDN access indicator set to ISDN_ACCES_IOBCI_INBAND then:	ISUP indicator parameter set to ISUP_ID, the
SIP Parameter values:		
ISUP Parameter	ACM FCI: ISUP_ID, ISDN_ACCESS_ID	
values:	OBCI: OBCI_INBAND;	
Comments:	SIP	JT ISUP
	INVITE →	→ IAM
	180 Ringing ←	<b>←</b> ACM
		g tone
	200 OK INVITE ←	<b>←</b> ANM
	ACK →	
		rsation
	BYE →	→ REL
	200 OK BYE <b>←</b>	<b>←</b> 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 Add	ress complete message (ACM)/	
SIP selection	PICS 2/1 AND PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:	, ,,	of BCI into PSTN XML ProgressIndicator #7 in the	
	sent 180		
	Ensure that the SLIT on receipt of an AC	CM message where the <b>CPS indicator</b> is set to	
		set to "ISUP is used all the way", the ISDN access	
	indicator is set to "ISDN":	set to 1001 is used all the way, the 10014 docess	
		with P-Early Media and <b>PSTN XML</b>	
	ProgressIndicator "Terminating		
SIP Parameter	180 Ringing response with P-Early Media and PSTN XML ProgressIndicator		
values:	"Terminating access ISDN"(#7).	-	
ISUP Parameter	ACM; CPS indicator: subscriber free		
values:	ISUP indicator: ISUP is used all the way	ay,	
	ISDN access indicator: ISDN		
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
	F	Ringing tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	C	Conversation	
	BYE →	→ REL	
	200 OK BYE <b>←</b>	<b>←</b> 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 c	omplete message (ACM)/	
SIP selection criteria:	PICS 2/1 AND PICS 4/18		
ISUP selection criteria:			
Test purpose:  SIP Parameter values:	ACM subscriber free and OBCI inband info available received, mapping 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.  180 Ringing P-Early Media and PSTN XML ProgressIndicator set to PI_ID.		
ISUP Parameter	ACM; CPS indicator: subscriber free		
values:	ISUP indicator: ISUP_ID		
	ISDN access indicator: ISDN_ACCES_ID		
Comments:	SIP SL		
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	Ringin	g tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conver	rsation	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> 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	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	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	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	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 <i>Terminating access ISDN"(#7)</i>	

TP104013	SIP reference: RFC 3261 [6]	ES 283 02 ETS 300 35	SUP reference: 27 [1], clause 7.2.3.1.4.1 6-1 [28], clauses 2.1.4, 2.2
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address	complete messa	ge (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 TMU_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).		
values:	PSTN XML first Bearer Capability: INVITE _BC1 PSTN XML second Bearer Capability: INVITE _BC2		
	180 Ringing PSTN XML BC: BC_VALUE	TE_BC2	
ISUP Parameter	ACM; CPS indicator: subscriber free,		
values:	ISUP indicator: ISUP is used all the way		
	ISDN access indicator: ISDN		
	TMU: TMU_VALUE		
	ATP: BC ATP_VALUE		
Comments:		UT	ISUP
	INVITE →	<b>→</b>	IAM
	180 Ringing ←	<b>+</b>	ACM
	200 OK INVITE	ng tone	ANM
	ACK	~	VIAIAI
	-	ersation	
	BYE →	→ ·	REL
	200 OK BYE	É	RLC

Values and selection criteria for test purpose TP104013			
Test	ACM Parameter values	180 Ringing Parameter	INVITE parameter value
purposes		values:	
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	
T00 (		ETS 300 356-1 [28], clause 2.1.4	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address of	complete message (ACM)/	
SIP selection criteria:	PICS 4/18		
ISUP selection	PICS 4/19		
criteria:			
Test purpose:	Fallback occurs in the early ACM mapping of PSTN XML ProgressIndicators  Ensure that the SUT on receipt of an ACM me	essage where the CPS indicator is set to "no	
	indication", the ISUP indicator is set to "ISUP indicator is set to "ISDN" and the Transmissio value TMU_VALUE and the BC in the Access ATP_VALUE:	n Medium Used (TMU) is included with the Transport Parameter (ATP) set to	
	ISDN"(#7).		
SIP Parameter	INVITE;		
values:	PSTN XML first Bearer Capability: INVITE _BC1		
	PSTN XML second Bearer Capability: INVIT		
ISUP Parameter	183 Session Progress; PSTN XML BearerCa	pability: ISDN_BC_VALUE	
values:	ACM; CPS indicator: no indication, ISUP indicator: ISUP is used all the way		
values.	ISDN access indicator: ISDN		
	TMU: TMU_VALUE		
	ATP: BC ATP_VALUE		
Comments:	SIP SL	JT ISUP	
	INVITE →	→ IAM	
	183 Session Progress ←	← ACM	
	180 Ringing ←	<b>←</b> CPG	
	Ringin	~	
	200 OK INVITE ←	← ANM	
	ACK →		
	Conver		
	BYE -	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

	Values and selection criteria for test purpose TP104014			
Test	ACM Parameter values	183 Session Progress	INVITE parameter value	
purposes		Parameter values:		
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	

TP104015	SIP reference: RFC 3261 [6]		clause 7.2.3.1.4A
			28], clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address of	omplete message (ACI	M)/
SIP selection	PICS 4/18		
criteria:	DIGG 4/40		
ISUP selection criteria:	PICS 4/19		
Test purpose:	Fallback occurs in the early ACM mapping of	LIC und DC in ATD int	to 102 DCTN VMI
rest purpose.	ProgressIndicators	ILC UNG BC III ATF IIIC	0 103 F3111 AIVIL
	Ensure that the SUT on receipt of an ACM message where the <b>CPS indicator</b> is set to "no indication", the <b>ISUP indicator</b> is set to "ISUP is used all the way", the <b>ISDN access indicator</b> is set to "ISDN" and containing an Access Transport Parameter (ATP) including <b>a High Layer Compatibility</b> (HLC) and containing the <b>progress indicator</b> #5: "interworking has occurred and has resulted in a telecommunication service change":		
	sends a 183 Session Progress messa the progress indication "interworking I telecommunication service change" (# the PSTN XML HighLayerCapability	as occurred and has re 5) " <i>Terminating access</i>	esulted in a
SIP Parameter	INVITE: <b>HLC</b> : HLC_VALUE1 (PIXIT), HLC_V		
values:	180 Session Progress; PSTN XML ProgressIndicator: interworking has occurred and has		
	resulted in a telecommunication service change (#5)  PSTN XML HighLayerCapability: HLC_VALUE2 (PIXIT)		
ISUP Parameter	ACM; CPS indicator: no indication,		
values:	ISUP indicator: ISUP is used all the way		
	ISDN access indicator: ISDN		
	ATP: progress indicator: interworking has o	curred and has resulte	ed in a
	telecommunication service change (#5)		
Comments:	HLC: HLC_VALUE2 (PIXIT)	T IOLID	
Comments:	SIP SU		
		2 17 1111	
	1.00 0.000.000	← ACM ← CPG	
	Ringin 200 OK INVITE ←	tone <b>←</b> ANM	
	ACK →	AINIVI	
	Conve	eation	
	BYE →	Salion → REL	
	200 OK BYE	← RLC	
	200 ON DIL	· NEC	

TP104016	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 of	ETS 300 356-1 [28], clause 2.1.4	
SIP selection	PICS 4/18	Complete message (AON)	
criteria:	1 103 4/10		
ISUP selection	PICS 4/19		
criteria:			
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 <b>CPS indicator</b> is set to "subscriber free", the <b>ISUP indicator</b> is set to "ISUP is used all the way", the <b>ISDN access indicator</b> is set to "ISDN" and containing an Access Transport Parameter (ATP) including a <b>High Layer Compatibility</b> (HLC) and containing the <b>progress indicator</b> #5: "interworking has occurred and has resulted in a telecommunication service change":		
	progress indication "interworking has telecommunication service change" (# the PSTN XML HighLayerCapability	#5) "Terminating access ISDN"(#7) and with	
SIP Parameter	INVITE: PSTN XML HighLayerCapability: H	ILC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT)	
values:	180 Ringing; PSTN XML ProgressIndicator: interworking has occurred and has resulted in a		
		has occurred and has resulted in a	
	telecommunication service change (#5) PSTN XML HighLayerCapability.: HLC_VAL	LIE2 (PIXIT)	
ISUP Parameter	ACM; CPS indicator: subscriber free,	1012 (17/11)	
values:	ISUP indicator: ISUP is used all the way		
	ISDN access indicator: ISDN		
	ATP: progress indicator: interworking has o	occurred and has resulted in a	
	telecommunication service change (#5)		
Commenter	HLC: HLC_VALUE2 (PIXIT)	LIT IOLID	
Comments:	SIP SU	UT ISUP → IAM	
	180 Ringing ←	← ACM	
		ng tone	
	200 OK INVITE ←	← ANM	
	ACK →	7 (141)	
		ersation	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> 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 c	omplete message (ACM)/
SIP selection criteria:	PICS 4/15 AND PICS 4/18	
ISUP selection criteria:		
Test purpose:	ACM free terminating access ISDN received. header	
	Ensure that SUT on receipt of an ACM messa "subscriber free" and if the I-MGCF has receiv request, and has not already sent a provisiona with parameters indicating authorization of earlier send the 180 Ringing response with media	red the P-Early-Media header in the INVITE all response including a P-Early-Media header
SIP Parameter	INVITE: P-Early-Media header, SDP audio xx	xxx RTP/AVP 8
values:	180 Ringing: P-Early-Media header	
ISUP Parameter values:	ACM; CPS indicator: free,	
Comments:	SIP SL	JT ISUP
	INVITE →	→ IAM
	180 Ringing ←	<b>←</b> ACM
	Ringin	<del>-</del>
	200 OK INVITE ← →	<b>←</b> ANM
	Conver	
	BYE →	→ REL
	200 OK BYE <b>←</b>	<b>←</b> 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 of			
SIP selection	PICS 4/18			
criteria:				
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:			
	<ul> <li>sends 180 response with a PSTN XML LowLayerCompatibility information element set to LLC_VALUE</li> </ul>			
SIP Parameter values:	180: PSTN XML LowLayerCompatibility: Ll	LC_VALUE (PIXIT)		
ISUP Parameter values:	ACM; ATP LLC: LLC_VALUE			
Comments:	SIP SI	JT ISUP		
	INVITE ->	→ IAM		
	180 Ringing ←	← ACM		
		g tone		
	200 OK INVITE ←	<b>←</b> ANM		
	ACK →			
		rsation		
	BYE 200 OK BYF  ←	→ REL ← RLC		
	200 OK BYE ←	₹ KLU		

TP104019	SIP reference: RFC 32	61 [6]	_	SUP reference: 027 [1], clause 7.2.3.1.4
				356-1 [28], clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt or	f the Address c		
SIP selection	PICS 4/18			
criteria:				
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 early 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	183: PSTN XML LowLayerCo	mpatibility: LL	C_VALUE (PI)	(IT)
values:			•	•
ISUP Parameter values:	ACM; <b>ATP LLC</b> : LLC_VALUE			
Comments:	SIP	SU	IT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	183 Session Progress	←	<b>←</b>	ACM
	180 Ringing	←	<b>←</b>	CPG
		Ringing	g tone	
		←	+	ANM
	ACK ·	<b>→</b>		
		Conver		
		<b>→</b>	<b>→</b>	REL
	200 OK BYE	€	+	RLC

## 6.2.1.5 Receipt of the Call progress message (CPG)

TP105001	SIP reference: R	FC 3261 [6]	_		ISUP reference:
					027 [1], clause 7.2.3.1.4
TSS reference:	SIP-ISUP/Basic call/ Rec	eipt of the Call	progress mess	sage (	CPG).
SIP selection	NOT PICS 4/15				
criteria:					
ISUP selection					
criteria:					
Test purpose:	CPG Alerting is mapped in	into a 180 ringi	ng		
					led party status indicator "no
	indication", on receipt of a	a CPG messag	e where the <b>ev</b>	ent ir	nformation is set to "Alerting":
	the 180 Ringing S	SIP response is	s sent.		
SIP Parameter					
values:					
ISUP Parameter	ACM: Called party st		tion"		
values:	CPG; event informa	tion: Alerting			
Comments:	SIP		SUT		ISUP
	INVITE	<b>→</b>		<b>→</b>	IAM
				<b>←</b>	ACM (no indication)
	180 Ringing	<b>←</b>		<b>←</b>	CPG (Alerting)
	3 3		Ringing tone		3/
	200 OK INVITE	<b>←</b>	· ····g···g · ····	<b>←</b>	ANM
	ACK	<b>→</b>		_	
	, core	=	Conversation		
	BYE	<b>→</b> `	Johnstallon	<b>→</b>	REL
	200 OK BYE	<del>-</del>		<del>-</del>	RLC
	ZUU UN DIE	~			RLU

TP105002	SIP reference: RFC	3261 [6]	ES 283	ISUP reference: 3 027 [1], clause 7.2.3.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt	of the Call pro	gress message	(CPG).
SIP selection criteria:	NOT PICS 4/15	·		
ISUP selection criteria:				
Test purpose:	CPG Progress is not interwo	rked		
		PG message v		alled party status indicator "no information is set to "Progress":
SIP Parameter				
values:				
ISUP Parameter	ACM: Called party status "ne			
values:	CPG; event information: P	rogress		
Comments:	SIP		SUT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
			+	ACM (no indication)
			<b>+</b>	CPG (Progress)
	200 OK INVITE	<b>←</b>	<b>+</b>	· ANM
	ACK	<b>→</b>		
		Cor	nversation	
	BYE	<b>→</b>	<b>→</b>	REL
	200 OK BYE	<b>←</b>	<del>(</del>	RLC

TP105003	SIP reference: RFC 3261	[6]		ISUP reference: 027 [1], clause 7.2.3.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the	ne Call progr	ess message (	CPG).
SIP selection criteria:	NOT PICS 4/15			
ISUP selection criteria:				
Test purpose:	CPG "in-band information or an a	ppropriate p	attern is now a	vailable" 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 <b>event information</b> is set to "in-band information or an appropriate pattern is now available":  • the CPG is not interworked.			
SIP Parameter				
values:				
ISUP Parameter	ACM: Called party status "no ind			
values:	CPG; event information: in-band	d-information	or an appropri	ate pattern is now available
Comments:	SIP	S	UT	ISUP
	INVITE	•	<b>→</b>	IAM
			+	ACM (no indication)
			<b>←</b>	CPG (Inband info)
	200 OK INVITE		<b>←</b>	ANM
		Conve	ersation	
	BYE -	•	<b>→</b>	REL
	200 OK BYE	•	+	RLC

TP105004	SIP reference: RFC	3261 [6]	ES 283 (	ISUP reference: 027 [1], clause 7.2.3.1.4A
TSS reference:	SIP-ISUP/Basic call/ Receipt	of the Call progr	ess 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 <b>event information</b> 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-Ea	rly-Media Heade	ſ	
ISUP Parameter values:	ACM: Called party status "no indication" CPG; event information: in-band-information or an appropriate pattern is now available			
Comments:	SIP INVITE		UT →	ISUP IAM ACM (no indication)
	183 Session Progress	<b>←</b>	<b>←</b>	CPG (Inband info)
	200 OK INVITE ACK	<b>←</b> →	<b>+</b> ersation	ANM
	BYE 200 OK BYE	→ +	ersation → ←	REL RLC

TP105005	SIP reference: RF	C 3261 [6]	FS 283	ISUP reference: 027 [1], clause 7.2.3.1.4
				0 356-1 [28], clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Recei	pt of the Call pro		
SIP selection	PICS 4/15 AND PICS 4/18			
criteria:				
ISUP selection				
criteria:				
Test purpose:	CPG Alerting received, 180	ocontaining a P-l	Early-Media head	der is sent
	Engure that the CLIT havin	a received the A	CM massage on	receipt of a CDC massage
	where the <b>event informati</b>			receipt of a CPG message
	sends an 180 Ring		•	included.
SIP Parameter	180 Ringing PSTN XML Pr			Media
values:	Too Kinging FOTT XIVE FF	ogressmaleator	i i_iD, i Laily i	vicula
ISUP Parameter	CPG; event information:	Alerting		
values:	,	3		
Comments:	SIP		SUT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	183 Session Progress	<b>←</b>	<b>←</b>	ACM (no indication)
		_	_	
	180 Ringing	<b>←</b>	<b>←</b>	CPG (Alerting)
		•	ging tone	
	200 OK INVITE	<b>←</b>	+	ANM
	ACK	<b>→</b>		
			versation	
	BYE	<b>→</b>	<b>→</b>	REL
	200 OK BYE	+	+	RLC

TP105006	SIP reference: F	RFC 3261 [6]		ISUP reference:	
				027 [1], clause 7.2.3.1.4	
				0 356-1 [28], clause 2.1.4	
TSS reference:	SIP-ISUP/Basic call/ Re	ceipt of the Call pro	gress message (	CPG).	
SIP selection	PICS 4/18				
criteria:					
ISUP selection criteria:					
Test purpose:	CPG Alerting contains a	n ATP with Progres	s Indicator		
	Ensure that SUT having received the ACM message, on receipt of a CPG message where the <b>event information</b> is set to "Alerting" and the <b>ATP</b> containing the <b>progress indicator</b> PI VALUE:				
	<ul> <li>sends an 180 Ringing response with PSTN XML body containing containing PSTN XML ProgressIndicator set to PI_VALUE.</li> </ul>				
SIP Parameter	180 Ringing PSTN XML	progress indicato	r PI_ID		
values:					
ISUP Parameter	CPG; event information: Alerting, ATP containing the ProgressIndicator PI_VALUE				
values:					
Comments:	SIP		SUT	ISUP	
	INVITE	<b>→</b>	→	IAM	
			+	ACM (no indication)	
	180 Ringing	<b>←</b> Ring	← ging tone	CPG (Alerting)	
	200 OK INVITE	<b>←</b>	<b>+</b>	ANM	
	ACK	<b>→</b>			
		Con	versation		
	BYE	<b>→</b>	<b>→</b>	REL	
	200 OK BYE	+	<b>←</b>	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	rogress message (CPG).
SIP selection criteria:	PICS 4/18	
ISUP selection criteria:		
SIP Parameter values: ISUP Parameter values:	where the <b>event information</b> is set to " A ISUP_ID, the <b>ISDN access indicator</b> se information set to OBCI_INBAND:	ACM message, on receipt of a CPG message slerting" the ISUP indicator parameter set to to ISDN_ACCES_ID and the OBCI in-band with P-Early Media and PSTN XML body containing mation element set to PI_ID.
Comments:	SIP INVITE →	SUT ISUP  → IAM  ← ACM (no indication)
	180 Ringing ←  200 OK INVITE ←  ACK →	<ul><li>← CPG (Alerting)</li><li>inging tone</li><li>← ANM</li></ul>
		onversation
	BYE → 200 OK BYE ←	→ REL ← RLC

	Values for test purposes TP105007				
test purposes	ISUP Parameter values:	PSTN XML progress descriptions:			
VA_01	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	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	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 (" <i>Terminating access ISDN"(#7))</i>			

TP105008	SIP reference: RFC 3261	[6]		ISUP reference: 027 [1], clause 7.2.3.1.4A	
				0 356-1 [28], clause 2.1.4	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the	ne Call progr	ess message (	CPG).	
SIP selection criteria:	PICS 4/18				
ISUP selection criteria:					
Test purpose:	CPG Progress containing a BCI a	and OBCI, m	apping into PS	TN XML instance	
	indicator set to PI_ID.	et to "Progre indicator s INBAND: gress with P	ess" and the <b>IS</b> et to CPG_ISD STN XML body	UP indicator parameter set to N_ACCESS_ID and the OBCI	
SIP Parameter values:	183 Session Progress PSTN XMI	<sub>-</sub> progress i	ndicator PI_IE		
ISUP Parameter	CPG; event information: Progress				
values:	ISUP indicator: CPG_ISUP_ID ISDN access indicator: CPG_IS	DN_ACCES	S_ID		
Comments:	SIP		UT	ISUP	
	INVITE -	•	<b>→</b>	IAM	
	183 Session Progress	-	<b>←</b>	ACM (no indication)	
	183 Session Progress		← ng tone	CPG (Progress)	
	200 OK INVITE	•	<b>←</b>	ANM	
	ACK -	•			
		Conve	ersation		
	BYE -	•	<b>→</b>	REL	
	200 OK BYE ★	•	<b>←</b>	RLC	

Values for test purposes TP105008					
Test purposes	ISUP Parameter values:	ISDN Parameter values:			
VA_01	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	PICS 4/15 AND PICS 4/18		
criteria:			
ISUP selection criteria:			
Test purpose:	available", the ISUP indicator set to CPG_ISUCPG_ISDN_ACCESS_ID and if the I-MGCF has received the P-Early-MGCF shall:	P-Early-Media header  M message, on receipt of a CPG message and information or an appropriate pattern is now	
SIP Parameter values:	183 Session Progress with P-Early Media and PSTN XML ProgressIndicator: PI_ID		
ISUP Parameter	CPG; event information: In-band information	or an appropriate pattern is now available	
values:	ISUP indicator: CPG_ISUP_ID		
	ISDN access indicator: CPG_ISDN_ACCES		
Comments:		UT ISUP	
	INVITE ->	→ IAM	
		← ACM (no indication)	
	183 Session Progress ←	← CPG (In-band info)	
	1	ng tone	
	200 OK INVITE ←	← ANM	
	ACK →		
		ersation	
	BYE →	→ REL	
	200 OK BYE <b>←</b>	<b>←</b> 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_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_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 32	61 [6]		ISUP reference: 27 [1], clause 7.2.3.2.1.4.1
				356-1 [28], clause 2.1.4
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).			
SIP selection	PICS 4/18			
criteria:				
ISUP selection criteria:	PICS 4/19			
Test purpose:	Fallback procedure: CPG Alert into 180	ing received, n	napping of TMU	and BC in the included ATP
	Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the event information 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	INVITE;			
values:	first Bearer Capability: INVITE_BC1 second Bearer Capability: INVITE_BC2 180 Ringing;			
ISUP Parameter	PSTN XML BearerCapability: IS CPG; event information: Alertin		NLUE .	
values:	TMU: TMU_VALUE	ig.		
raidoi	ATP: BC ATP_VALUE			
Comments:	SIP	S	UT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
			<b>←</b>	ACM (no indication)
	180 Ringing	<b>←</b>	<b>+</b>	CPG (Alerting)
	200 OK INVITE	€	ng tone	ANM
	ACK	<b>→</b>	•	, , , , , ,
		Conve	ersation	
	BYE	<b>→</b>	<b>→</b>	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 C	all progress r	nessage (CPG).
SIP selection	PICS 4/18		
criteria:			
ISUP selection	PICS 4/19		
criteria:			
Test purpose:	into a 183		ping of TMU and BC in the included ATP
	where the <b>event information</b> is set to requirement is included with the value Parameter is set to ATP_VALUE:  • sends a 183 Session Progres	o "Progress": e TMU_VAL	
	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: Progress TMU: TMU_VALUE ATP: BC: ATP_VALUE		
Comments:	SIP	SUT	ISUP
	INVITE ->		→ IAM
			← ACM (no indication)
	183 Session Progress ←	Ringing to	← CPG (Progress)
	200 OK INVITE ←		← ANM
	ACK →		- / 11 1111
	,,,,,,	Conversati	on
	BYE →	3011V01341	→ REL
	200 OK BYE		← RLC
	ZOO ON DIE		- INLO

	Values and selection c	riteria for test purposes TP1	05010 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:
TSS reference:	SID ISLID/Pagin call/ Pagaint of th	o Call progr		27 [1], clause 7.2.3.2.1.4.1
SIP selection	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).  PICS 4/18			
criteria:	1 103 4/10			
ISUP selection				
criteria:				
Test purpose:	CPG Alerting received, sending o and HLC in the 180	f received Pi	#5 and HLC ir	n an ATP into a PSTN XML PI
	Ensure that the SUT, having rece where the <b>event information</b> is s Parameter including a <b>High Laye indicator</b> #5: "interworking has ochange":	et to " Alertir r Compatibi	ng″and contain <b>lity</b> (HLC) and	ing an Access Transport containing the <b>progress</b>
	<ul> <li>sends an 180 Ringing me progress indication "interv telecommunication servic HighLayerCapability.</li> </ul>	vorking has	occurred and h	
SIP Parameter	INVITE:			
values:	PSTN XML HighLayerCompatibility: HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT)			
	180 Ringing;			and become without the co
	<b>PSTN XML ProgressIndicator</b> : in telecommunication service change		nas occurred a	nd has resulted in a
	PSTN XML HighLayerCapability		IF2 (PIXIT)	
ISUP Parameter	CPG; event information: Alerting		)	
values:	progress indicator: interworking has occurred and has resulted in a telecommunication			
	service change (#5)			
	HLC: HLC_VALUE2 (PIXIT)			
Comments:	SIP		JT	ISUP
	INVITE -	•	<b>→</b>	IAM
			+	ACM (no indication)
	180 Ringing		<b>←</b>	CPG (Alerting)
	Too Kinging		g tone	or o (Merung)
	200 OK INVITE	•	<b>←</b>	ANM
	ACK ->			
		Conve	rsation	
	BYE →	•	<b>→</b>	REL
	200 OK BYE	ı	+	RLC

TP105013	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 prog		
SIP selection	PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:	CPG Progress received, sending of received and HLC in the 183	PI #5 and HLC in an ATP in a PSTN XML PI	
	Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the <b>event information</b> is set to "Progress" and containing an Access Transport Parameter (ATP) including a <b>High Layer Compatibility</b> (HLC) and containing the <b>progress indicator</b> #5: "interworking has occurred and has resulted in a telecommunication service change":		
	the progress indication "interworking telecommunication service change" HighLayerCapability.	(#5) and with the <b>PSTN XML</b>	
SIP Parameter	INVITE: PSTN XML HighLayerCapability: HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT)		
values:	183 Session Progress; PSTN XML ProgressIndicator: interworking has occurred and has		
	resulted in a telecommunication service change (#5)		
10110.0	PSTN XML HighLayerCapability: HLC_VALUE2 (PIXIT)		
ISUP Parameter	CPG; event information: Progress  ATP: progress indicator: interworking has occurred and has resulted in a		
values:	telecommunication service change (#5) <b>High Layer Capability</b> : HLC_VALUE2 (PIXIT)		
Comments:		SUT ISUP	
Gommonto.	INVITE →	→ IAM	
		← ACM (no indication)	
		,	
	183 Session Progress ←	← CPG (Progress)	
	Ring	ing tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
		ersation	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

TP105014	SIP reference: RFC 3261 [6	5]		ISUP reference: 27 [1], clause 7.2.3.2.1.4A
TSS reference:	SIP-ISUP/Basic call/ Receipt of the	Call progre	ess message (0	CPG).
SIP selection criteria:	PICS 4/18			
ISUP selection criteria:				
SIP Parameter values: ISUP Parameter values:	CPG Progress containing ATP PI, m 183  Ensure that the SUT having received where the event information is set set to PI-VALUE, the BCI ISUP indicator set to ISUP indicator set to ISUP indicator set to ISUP information element set to PI-SUN" (#7)  ACM; BCI ISUP indicator: ISUP using CPG: event information: Progress	d the ACM to "Progres cator para SDN: ess messa I_ID. ProgressIr	I message, on ss", the ATP commeter set to IS ge with the PS dicator: PI_ID a	receipt of a CPG message ontains the progress indicator SUP used all the way and the
	CPG; event information: Progress BCI ISUP indicator: ISUP used all t BCI ISDN access indicator: ISDN	he way		
	ATP Progress Indicator value PI_ID			
Comments:	SIP INVITE →	SI	JT → ←	ISUP IAM ACM (no indication)
	183 Session Progress ←	Ringin	<b>←</b> g tone	CPG (Progress)
	200 OK INVITE ← ACK →		<b>←</b>	ANM
		Conve	rsation	
	BYE → 200 OK BYE ←		<b>→</b>	REL RLC

Values and additional selection criteria for test purpose TP105014		
VA_01	PI_ID = Call is not end-to-end ISDN (#1)	
VA_02 PI_ID = 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 Answ	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	ANM received, a 200 OK INVITE is sent	
	Ensure that the SUT, having received the "subscriber free", on receipt of an ANM m  sends a 200 OK INVITE to the U.	, and the second
SIP Parameter		
values:		
ISUP Parameter values:		
Comments:	SIP SU	T ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM (free)
	F	Ringing tone
	200 OK INVITE ←	← ANM
	ACK →	
	C	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			
SIP selection criteria:	PICS 4/18	-		
ISUP selection criteria:				
Test purpose:	ANM received, mapping of PI contained in the	ne ATP into the 200 OK PSTN XML PI		
SIP Parameter	Ensure that the SUT, having received the ACM message, on receipt of an ANM message containing a <b>progress indicator</b> set to PI_VALUE in the ATP:  • sends a 200 OK included the <b>PSTN XML ProgressIndicator</b> set to PI_VALUE.			
values:	200 OK, PSTN AML Progressingicator. Pi	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)		
		ing tone		
	200 OK INVITE ←	← ANM		
	ACK →			
	Conv	versation versation		
	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	er message (ANM).	
SIP selection	PICS 4/18		
criteria:			
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		
	Capability (BC):	set to TMU_VALUE and the ATP without Bearer PSTN XML Bearer Capability encoded	
SIP Parameter	INVITE:		
values:	PSTN XML first Bearer Capability: SETUP_BC1		
	PSTN XML second Bearer Capability: SETUP_BC2		
	200 OK; Bearer capability: ISDN_BC_VAL	UE	
ISUP Parameter	ANM; TMU: TMU_VALUE		
values:	ATP: no BC		
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM (free)	
	R	nging tone	
	200 OK INVITE ←	← ANM	
	ACK →		
	Co	nversation	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

Values for test purposes TP106003				
Test purposes	ANM parameter values	200 OK parameter values	SETUP parameter values	
/A_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]			SUP reference: 027 [1], clause 7.2.3.1.5
				356-1 [28], clause 2.1.7
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).			
SIP selection	PICS 4/18		,	
criteria:				
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 containing the <b>Transmission Medium</b> ATP_VALUE and containing the <b>progr</b>	Used set	to TMŬ_VAL	UE and the ATP set to
	has resulted in a telecommunication se			nerworking has occurred and
	<ul> <li>sends a 200 OK message with ISDN_BC_VALUE and the PS occurred and has resulted in a</li> </ul>	TN XML P	rogressIndi	cator set to "interworking has
SIP Parameter	INVITE:	101000111111	unioation coi	vice onange (ne).
values:	PSTN XML first BearerCapability: SETUP_BC1			
	PSTN XML second Bearer Capability: SETUP_BC2			
	200 OK;			
	PSTN XML BearerCapability: ISDN_BC_VALUE			
	<b>ProgressIndication</b> : interworking has occurred and has resulted in a telecommunication			
IOUD Developed	service change(#5)			
ISUP Parameter values:	ANM; <b>TMU</b> : TMU_VALUE ATP: ATP_VALUE			
values.	Progress indication: interworking has	s occurred :	and has resi	ilted in a telecommunication
	service change(#5)	o o o o o o o o o o o o o o o o o o o	a	
Comments:	SIP	SUT		ISUP
	INVITE ->		<b>→</b>	IAM
	180 Ringing ←		<b>←</b>	ACM (free)
		Ringing to	one	,
	200 OK INVITE ←		<b>←</b>	ANM
	ACK →			
		Conversa	tion	
	BYE →		<b>→</b>	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 <b>ProgressIndicator</b> : (#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 <b>ProgressIndicator</b> : (#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 m	essage (ANM).	
SIP selection	PICS 4/18		
criteria:			
ISUP selection criteria:	PICS 4/19		
Test purpose:	Fallback procedure: ANM received contains a	n ATP with BC unrestricted digital information	
	with tones/announcement, mapping into the F	PSTN XML PI in the sent 200 OK INVITE	
	Francisco that the CUT having received the AC	M. manager and manager of an ANIM manager	
	Ensure that the SUT, having received the AC containing the ATP including the <b>Bearer Cap</b> .		
	with tones/announcement" and without <b>TMU</b>		
	with tones/announcement and without <b>two</b> parameter.		
	sends a 200 OK message with the PSTN XML Bearer Capability set to		
	"unrestricted digital information with to		
SIP Parameter	200 OK; PSTN XML BearerCapabilty: unres	tricted digital information with	
values:	tones/announcement		
ISUP Parameter	ANM; ATP BC: unrestricted digital information	n with tones/announcement	
values:	no TMU		
Comments:		SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM (free)	
		ng tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →	_	
		ersation	
	BYE →	→ REL	
	200 OK BYE <b>←</b>	<b>←</b> RLC	

TP106006	SIP reference: RFC 3261 [6]			ISUP reference:
				027 [1], clause 7.2.3.1.5
		E1	S 300	) 356-1 [28], clause 2.1.7
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Ansv	ver message (A	ANM).	
SIP selection	PICS 4/18			
criteria:				
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 th			
	containing the HLC parameter in the AT	<b>P</b> set to HLC_\	/ALUE	<u>:</u> :
	sends a 200 OK message PSTN	I XML HighLay	yerCo	mpatibility information
	element set to HLC_VALUE.			
SIP Parameter	200 OK;			
values:	PSTN XML HighLayerCompatibility: HLC_VALUE (PIXIT)			
ISUP Parameter	ANM; ATP HLC: HLC_VALUE			
values:				
Comments:	SIP	SUT	_	ISUP
	INVITE →		<b>→</b>	IAM
	180 Ringing ←		<b>←</b>	ACM (free)
		Ringing tone		
	200 OK INVITE ←		<b>←</b>	ANM
	ACK →			
		Conversation		
	BYE →		<b>→</b>	REL
	200 OK BYE ←		+	RLC

TP106007	SIP reference: RFC 3261 [6]	ES 283 ETS 300	ISUP reference: 027 [1], clause 7.2.3.1.5 ) 356-1 [28], clause 2.1.7	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer r	nessage (ANM).		
SIP selection criteria:	PICS 4/18			
ISUP selection criteria:	PICS 4/19			
SIP Parameter values:	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).  INVITE: PSTN XML HighLayerCompatibility: HLC_VALUE1 (PIXIT), HLC_VALUE2 (PIXIT) 200 OK; PSTN XML HighLayerCompatibility: HLC_VALUE2 (PIXIT)			
	PSTN XML ProgressIndicator: interworking has occurred and has resulted in a			
ICUD D	telecommunication service change (#5)			
ISUP Parameter	ANM; ATP HLC: HLC_VALUE2	ad and has record	Itad in a talagammunigation	
values:	progress indicator: interworking has occurred and has resulted in a telecommunication			
Comments:	service change (#5)	SUT	ISUP	
Comments.	INVITE →	→	IAM	
	180 Ringing	<b>É</b>	ACM (free)	
		ing tone	, to w (1100)	
	200 OK INVITE	<b>←</b>	ANM	
	ACK →			
	Conv	ersation		
	BYE →	<b>→</b>	REL	
	200 OK BYE <b>←</b>	+	RLC	

TP106008	SIP reference: RFC 326	1 [6]		ISUP reference: 027 [1], clause 7.2.3.1.5 0 356-1 [28], clause 2.1.7
TSS reference:	SIP-ISUP/Basic call/ Receipt of t	he Answer m	essage (ANM).	
SIP selection	PICS 4/18			
criteria:				
ISUP selection criteria:				
Test purpose:	ANM received contains an ATP of sent 200 OK INVITE  Ensure that the SUT, having recontaining the LLC parameter in  sends a 200 OK message element set to LLC_VAL 200 OK INVITE: PSTN XML Low	eived the ACI the <b>ATP</b> set ge with a <b>PST</b> UE.	M message, on to LLC_VALUE	receipt of an ANM message : yerCompatibility information
values:				
ISUP Parameter values:	ANM; ATP LLC: LLC_VALUE			
Comments:	SIP	S	UT	ISUP
	INVITE	<b>→</b>	→	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM (free)
		Ringir	ng tone	
	200 OK INVITE	←	<b>←</b>	ANM
	ACK	<b>→</b>		
		Conve	ersation	
	BYE	<b>→</b>	→	REL
	200 OK BYE	←	<b>←</b>	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 CONNEC			
SIP selection criteria:	PICS 4/1 AND PICS 4/4 AND PICS 4/5			
ISUP selection criteria:				
Test purpose:	reconditions are met			
	SDP offer was received in the initial INVITE. message: sends a 200 OK INVITE to the UAC. The bearer path shall be connected in both directions.			
	The bearer path shall be connected in both directions when both of the following conditions are satisfied:			
	the I-MGCF determines (using the pro- sufficient preconditions have been sat establishment to proceed (if applicable)			
	bearer set-up in the forward direction" nnect Type is "notification not required", the ons when the Bearer Set-up request is sent lures defined in RFC 3312 [7]) that sufficient o proceed.			
SIP Parameter	INVITE: Require: precondition	•		
values:	SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sendes:qos none remote sendre			
	183: Require: 100rel SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory local sene a=conf:qos remote sendrecv			
	UPDATE: SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory local sendrecy a=des:qos mandatory remote se			
	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 se			
ISUP Parameter values:				
Comments:		UT ISUP		
	INVITE → 183 Session Progress ←	→ IAM		
	183 Session Progress ← PRACK →			
	200 OK PRACK ←			
	UPDATE →	→ COT(successful)		
	200 OK UPDATE ← 200 OK INVITE ←	← CON		
	200 OK INVITE ← ACK →	CON		
	-	ersation		
	BYE →	→ REL		
	200 OK BYE <b>←</b>	<b>←</b> 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 CONNEC			
SIP selection	NOT PICS 4/1 AND PICS 4/4 AND PICS 4/5			
criteria:				
ISUP selection criteria:				
Test purpose:	CON received, 200 OK INVITE is sent after Preconditions are met			
	<b>SDP offer was received</b> 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:			
	<ul> <li>the I-MGCF determines (using the pro sufficient preconditions have been sat establishment to proceed (if applicable)</li> </ul>	sfied on the SIP side for session		
	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	INVITE: Require: precondition			
values:	SDP a=curr:qos local none			
	a=curr:qos remote none	dracy		
	a=des:qos mandatory local sendrecv a=des:qos none remote sendrecv			
	a-ues.qus none remote senurecy			
	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 se	enarecv		
	200 OK UPDATE			
	SDP a=curr:qos local sendrecv			
	a=curr:qos remote sendrecv a=des:qos mandatory local send	drecy		
	a=des:qos mandatory rocal serio			
ISUP Parameter				
values:	CID	IT IOUE		
Comments:	SIP SI	JT ISUP		
	183 Session Progress			
	PRACK →			
	200 OK PRACK ←	-		
	UPDATE →	→ IAM		
	200 OK UPDATE ← 200 OK INVITE ←	<b>←</b> CON		
	ACK →	2 0014		
	Conversation			
	BYE →	→ REL		
	200 OK BYE ←	<b>←</b> 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		
	sends a 200 OK INVITE to the UAC. The bearer path shall be connected in both direction displayed by the series of the UAC. The bearer path shall be connected in both direction displayed by the series of the UAC.  In addition, if BICC is performing the "Per-call Outgoing bearer set-up procedure and the Cobearer path shall be connected in both direction did the I-IWU determines (through the procedure preconditions have been met for the session to	bearer set-up in the forward direction" nnect Type is "notification not required", the ons when the Bearer Set-up request is sent lures defined in RFC 3312 [7]) that sufficient	
SIP Parameter	INVITE: SDP offer		
values:	200 OK INVITE: SDP answer		
ISUP Parameter values:			
Comments:	SIP S	UT ISUP	
	INVITE →	→ IAM	
	200 OK INVITE ←	← CON	
	ACK →		
		ersation	
	BYE →	→ REL	
	200 OK BYE <b>←</b>	<b>←</b> 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	PICS 4/18	
criteria:		
ISUP selection		
criteria:		
Test purpose:	CON received contains a PI conveyed in al 200 OK INVITE	a ATP, mapped into the PSTN XML PI sent in the
	PI_VALUE in the ATP:  • sends a 200 OK included the PSTN	message containing a <b>progress indicator</b> set to <b>IXML ProgressIndicator</b> 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 →	
	Cor	versation
	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 m	essage (CON).	
SIP selection	PICS 4/18		
criteria:			
ISUP selection criteria:	PICS 4/19		
Test purpose:	CON received contains the TMU parameter, n 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 <b>PSTN XML BearerCapability</b> encoded		
	ISDN_BC_VALUE.		
SIP Parameter	INVITE:		
values:	PSTN XML first Bearer Capability: SETUP_BC1		
	PSTN XML second Bearer Capability: SETU	JP_BC2	
	200 OK;		
ISUP Parameter	Bearer capability: ISDN_BC_VALUE CON: TMU: TMU VALUE		
values:	ATP: no BC		
Comments:		UT ISUP	
Gommonto.	INVITE →	→ IAM	
	200 OK INVITE ←	← CON	
	ACK →	- 33	
		ersation	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> 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	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer m	ETS 300 356-1 [28], clause 2.1.7	
SIP selection	PICS 4/18	0000g0 (0014).	
criteria:			
ISUP selection	PICS 4/19		
criteria:			
Test purpose:	CON received contains the TMU parameter a		
	mapping into the PSTN XML PI sent in the 20	0 OK INVITE	
	Ensure that the SUT , on receipt of an CON m	pessage containing the <b>Transmission</b>	
	<b>Medium Used</b> set to TMU_VALUE and the <b>A</b>		
	progress indicator set to "interworking has o		
	telecommunication service change" (#5):		
	<ul> <li>sends a 200 OK message with the PS</li> </ul>		
		L ProgressIndicator set to "interworking has	
OID D	occurred and has resulted in a telecor	mmunication service change" (#5).	
SIP Parameter values:	INVITE: PSTN XML first BearerCapability: SETUP_BC1		
values:	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	CON; <b>TMU</b> : TMU_VALUE		
values:	ATP: BC ATP_VALUE		
	<b>Progress indication</b> : interworking has occurred and has resulted in a telecommunication		
Comments:	service change(#5)	UT ISUP	
Comments.	INVITE →	→ IAM	
	200 OK INVITE	← CON	
	ACK →	CON	
	7.6.1	ersation	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

	Val	ues for test purposes TP107000	6
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 <b>ProgressIndicator</b> : (#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 <b>ProgressIndicator</b> : (#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 m	essage (CON).	
SIP selection	PICS 4/18		
criteria:			
ISUP selection	PICS 4/19		
criteria:			
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".  200 OK; PSTN XML BearerCapabilty: unrestricted digital information with		
values:	tones/announcement		
ISUP Parameter	CON; ATP BC: unrestricted digital information with tones/announcement		
values: Comments:	no TMU		
Comments:	INVITE →	-> 1004	
		→ IAM ← CON	
	200 OK INVITE	← CON	
	ACK →	a wa aki a wa	
		ersation	
	BYE -	→ REL	
	200 OK BYE <b>←</b>	<b>←</b> 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 m	nessage (CON).
SIP selection criteria:	PICS 4/18		
ISUP selection			
criteria:			
Test purpose:	CON received contains the ATP pa	arameter co	onveying the HLC, mapping into the PSTN
	XML HLC sent in the 200 OK INVI	TE	
	Ensure that the SUT, on receipt of	an CON m	essage containing the HLC parameter in the
	ATP set to HLC_VALUE:		
	<ul> <li>sends a 200 OK message PSTN XML HighLayerCompatibility information</li> </ul>		
	element set to HLC_VALUE.		
SIP Parameter	200 OK;		
values:	PSTN XML HighLayerCompatibility: HLC_VALUE (PIXIT)		
ISUP Parameter	CON; ATP HLC: HLC_VALUE		
values:			
Comments:	SIP	S	SUT ISUP
	INVITE →		→ IAM
	200 OK INVITE ←		← CON
	ACK →		
		Conve	ersation
	BYE →		→ REL
	200 OK BYE		<b>←</b> RLC

TP107009	SIP reference: RFC 3261 [6]	ES 283 02	UP reference: 27 [1], clause 7.2.3.1.5 56-1 [28], clause 2.1.7
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer		30-1 [20], Clause 2.1.7
SIP selection	PICS 4/18		
criteria:			
ISUP selection criteria:	PICS 4/19		
Test purpose:	CON received contains the ATP parameter	conveying the HI C	and PI #5 manning into the
SIP Parameter values:	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).  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	CON; ATP HLC: HLC_VALUE2		
values:	progress indicator: interworking has occurred and has resulted in a telecommunication		
	service change (#5)		
Comments:	SIP		SUP
	INVITE →		AM
	200 OK INVITE ←	← (	CON
	ACK →		
		versation	
	BYE →		REL
	200 OK BYE <b>←</b>	<b>←</b> F	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 m	essage (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:	INVITE  200 OK INVITE  ACK  →	UT ISUP  → IAM  ← CON  ersation  → REL  ← 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		
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		
SIP Parameter values:	header field.  cause value: CV_SIP (PIXIT)		
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)		
Comments:	SIP SU INVITE → SIP_FAILURE_VA ← ACK	JT ISUP  → IAM  ← REL → 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 ("call 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	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 to 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]		SUP reference:
			ES 283 (	)27 [1], clause 7.2.3.1.8
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/			
SIP selection	PICS 4/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	REL after ACM received, ma	pping in a final r	esponse	
	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	<u>-</u> )		
ISUP Parameter values:	REL; cause value: CV_ISUF	P (PIXIT)		
Comments:	SIP	SI	JT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
			<b>←</b>	ACM (no indication)
	SIP_FAILURE_VA	<b>←</b>	<b>←</b>	REL ,
	ACK	<b>→</b>	<b>→</b>	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 to 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:		g. (	
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		
SIP Parameter	header field.  Cause value: CV_SIP (PIXIT)		
values:	DEL Como del COVI ICUID (DIVIT)		
ISUP Parameter values:	REL; Cause value: CV_ISUP (PIXIT)		
Comments:	SIP SL		
	INVITE →	→ IAM	
	180 Ringing ←	← ACM (free)	
	SIP_FAILURE_VA ←	<b>←</b> 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	NOT PICS 4/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	REL received in the early dialogue (CPG Aler	ting) 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	REL; cause value: CV_ISUP (PIXIT)			
values:	, , , , , , , , , , , , , , , , , , , ,			
Comments:	SIP SL	IT ISUP		
	INVITE ->	→ IAM		
		<ul> <li>ACM (no indication)</li> </ul>		
	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 to 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	Cit 1001 / Badio daii/ Receipt of the Release I	neodago (NE <i>E)i</i>		
criteria:				
ISUP selection				
criteria:				
Test purpose:	REL received in the confirmed state (ANM rec	ceived)		
SIP Parameter	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.			
values:	Cause value: CV_SIP (PIXIT)			
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)			
Comments:	SIP SL	JT ISUP		
	INVITE →	→ IAM		
	180 Ringing ← ← ACM			
	200 OK INVITE ←	← ANM		
	ACK →			
	Conversation			
	BYE ←	<b>←</b> 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:	<ul> <li>REL received in the confirmed state (CON received)</li> <li>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 ISUP REL where the cause value defined as CV_ISUP:</li> <li>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;</li> <li>the SUT shall send BYE message.</li> <li>the ISUP Cause Value field in the ISUP REL message is mapped to the Reason</li> </ul>		
SIP Parameter	header field.  Cause value: CV SIP (PIXIT)		
values:	Cause value. CV_SIF (FIXIT)		
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)		
Comments:	SIP SU	JT ISUP	
	INVITE →	→ IAM	
	200 OK INVITE ←	← CON	
	ACK →		
	Conversation		
	BYE ←	<b>←</b> 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			
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 S INVITE → 484 Address incomplete ← ACK →	UT ISUP		

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			
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	480 is sent if congestion in the SUT			
SIP Parameter values:	Ensure that the SUT in congestion on receipt of INVITE message:     sends an 480 Temporarily Unavailable message.			
ISUP Parameter values:				
Comments:	SIP	SUT ISUP		
	INVITE ->			
	480 Temporarily unavailable ← ACK →			
	AUN 7			

TP109003	SIP reference: RFC 3261 [6]		ISUP reference: 27 [1], clause 7.2.3.1.8.10	
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I	1	2. [1], olddo 1.2.0110110	
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	500 is sent to due the compatibility procedure	for unknown pa	arameters	
	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 →	<b>→</b>	IAM	
		<del>(</del>	ACM (???)	
	500 Server internal error	<b>→</b>	REL	
	ACK →	<u> </u>	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 SI	UT ISUP		
	INVITE →	→ IAM		
	Expir	y of T7		
	484 Address incomplete ←	→ REL		
	ACK →	<b>←</b> 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	ase at I-l	-MGCF
SIP selection criteria:			
ISUP selection criteria:	PICS 4/16		
Test purpose:	Call setup is cleared after T9 expiry		
SIP Parameter values:	ensure that the call is released due exp     sends 480 Temporarily Unava	. ,	T9 within the BICC/ISUP procedures:
ISUP Parameter			
values:			
Comments:	SIP	SL	UT ISUP
	INVITE →		→ IAM
	180 Ringing ←		← ACM
		Expiry	y 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 compatibi	lity "Release call"	
Comments:	SIP S	UT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
		<b>←</b> XXX	
	500 Server internal error ←	→ REL	
	ACK →	<b>←</b> 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	messa	ige
SIP selection			
criteria:			
ISUP selection criteria:			
Test purpose:	BYE with Reason header received, send	ling of F	REL
	Ensure that the SUT on receipt of SIP B' side:	YE, the	SUT shall send an ISUP REL to the ISUP
	Value <b>is included</b> in the BYE n the ISUP REL message with the	nessage e locatio	rith ITU-T Recommendation Q.850 [5] Cause e is mapped to the ISUP Cause Value field in on "network beyond interworking point".
SIP Parameter values:	Protocol-cause: CV_Reason Header (Pl	XIT)	
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT)		
Comments:	SIP	SU	T 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 r	:
SIP selection	on real /Basic can, receipt of the c/ trope	meddagd
criteria:		
ISUP selection		
criteria:		
Test purpose:	CANCEL with Reason header received, sendi	ing of REL
	<ul> <li>Ensure that the Reason header field value is included in the CANCEL me</li> </ul>	EL, the I-MGCF shall send an ISUP REL to the with ITU-T Recommendation Q.850 [5] Cause essage is mapped to the ISUP Cause Value the location "network beyond interworking
SIP Parameter values:		
ISUP Parameter	REL: cause value: CV_ISUP (PIXIT)	
values:	location: "network beyond interworking point"	
Comments:	SIP	UT ISUP
	INVITE →	→ IAM
	180 Ringing ←	<b>←</b> ACM
	CANCEL -	→ REL
	200 OK CANCEL ←	← RLC
	487 Request Terminated ←	
	ACK →	

TP110003	SIP reference: RFC 32	261 [6]	50.000	ISUP reference:
			ES 283	3 027 [1], clause 7.2.3.1.6
TSS reference:	SIP-ISUP/Basic call/ Receipt of	f the BYE mes	sage	
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	BYE without Reason header re	eceived, sendin	g of REL	
	Ensure that the SUT on receip	t of SIP BYE w	ithout Reason I	header, the SUT shall send an
	ISUP REL to the ISUP side.			
	Ensure that the coding of the IS	SUP Cause Va	lue is # 16 with	the location "network beyond
	interworking point" if no reason	header is con	tained in the SI	P message.
SIP Parameter				
values:				
ISUP Parameter	REL: cause value: #16			
values:				
Comments:	SIP	S	UT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
	BYE	<b>→</b>	<b>→</b>	REL
	200 OK BYE	<del>-</del>	+	RLC

TP110004	SIP reference: RFC 3261 [6]		ISUP reference: 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, se	ending of REL	
	Ensure that the SUT on receipt of SIP CANCE an ISUP REL to the ISUP side.  Ensure that the coding of the ISUP Cause Valinterworking point" if no reason header is con	lue is # 31 with	the location "network beyond
SIP Parameter values:			
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT) location: LOC ISUP (PIXIT)		
Comments:	= \ /	SUT	ISUP
	INVITE ->	<b>→</b>	IAM
	180 Ringing ←	<b>←</b>	ACM
	CANCEL →	<b>→</b>	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 (CG	B) with the indication hardware failure oriented	
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	RSC received in the confirmed state, a BYE is	s 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 →		
	Conv	ersation	
	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 r		
	(GRS) or Circuit group blocking message (CG	B) with the indication hardware failure oriented	
SIP selection			
criteria:			
ISUP selection criteria:			
Test purpose:	GRS received in the confirmed state, a BYE is	s sent	
	,	n is in the confirmed state, on receipt of a GRS	
	message sends:		
	a RVE massage if the SLIT has alrea	dy received an ACK for the 200 OK INVITE	
	<ul> <li>a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent.</li> </ul>		
SIP Parameter			
values:			
ISUP Parameter			
values:			
Comments:		SUT ISUP	
	INVITE ->	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conv	rersation	
	BYE ←	<b>←</b> GRS	
	200 OK BYE →	→ GRA	

TP111003	SIP reference: RFC 3261 [6]	ISUP reference:	
<b></b>		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 (CG	B) with the indication hardware failure oriented	
SIP selection criteria:			
ISUP selection			
criteria:			
Test purpose:	CGB "Hardware failure oriented" received in the	ne confirmed state, a BYE is sent	
SIP Parameter	message, with the Circuit Group Supervision Mailure oriented", sends:	n is in the confirmed state, on receipt of a CGB Message Type Indicator coded as "hardware  dy received an ACK for the 200 OK INVITE	
values:			
ISUP Parameter	Circuit Group Supervision Message Type Indicator "hardware failure oriented"		
values:			
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conv	ersation	
	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 se	nt and ACK is not received	
		ceived, on receipt of a RSC message sends	
	200 OK INVITE if the SUT has not yet rece	ived 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 <b>←</b>		
	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 n (GRS) or Circuit group blocking message (CG	
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 receive 200 OK INVITE if the SUT has not yet receive	
	<ul> <li>The SUT shall wait until it receives the the BYE.</li> </ul>	e ACK for the 200 OK INVITE before sending
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP   S   S   S   S   S   S   S   S   S	SUT ISUP  → IAM  ← ACM ← ANM ← GRS → GRA

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 afte received	r 200 OK INVITE was sent and ACK is not		
	Ensure that the SUT, when an ANM was received, on receipt of a CGB 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	Circuit Group Supervision Message Type Indicator "hardware failure oriented"			
values:				
Comments:		UT ISUP		
	INVITE -	→ IAM		
	180 Ringing ←	← ACM		
	200 OK INVITE ←	<b>←</b> ANM		
		<b>←</b> CGB		
	ACK →	→ CGBA		
	BYE ←			
	200 OK BYE →			

TP111007	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 (CG	GB) with the indication hardware failure oriented	
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	RSC in the early dialogue received, a 500 is s	sent	
		vard ISUP/BICC message relating to the call has	
	already been received on receipt of a RSC message sends:		
	<ul> <li>a 480 Temporarily on the SIP side.</li> </ul>		
SIP Parameter			
values:			
ISUP Parameter			
values:			
Comments:	SIP	SUT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
		← RSC	
	480 Temporarily Unavailable ←	→ RLC	
	ACK →		

TP111008	SIP reference: RFC 326	61 [6]		ISUP reference: )27 [1], clause 7.2.3.1.8.9
TSS reference:	SIP-ISUP/Basic call/ Receipt of (GRS) or Circuit group blocking		message (RSC),	Circuit group reset message
SIP selection criteria:	(cree, or or or or group areasiming			
ISUP selection criteria:				
Test purpose:	GRS in the early dialogue received	ved, a 500 is s	sent	
	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 480 Temporarily Unavailable on the SIP side.			
SIP Parameter				
values: ISUP Parameter				
values:				
Comments:	SIP	•	SUT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM
			<b>←</b>	GRS
	480 Temporarily Unavailable	<b>←</b>	<b>→</b>	GRA
	ACK	<b>→</b>		

TP111009	SIP reference: RFC 32	61 [6]		ISUP reference: 127 [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	d" in the early o	lialogue received	d, 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 480 Temporarily Unavailable on the SIP side.			
SIP Parameter values:				
ISUP Parameter values:				
Comments:	SIP	5	SUT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM
	100 T	-	<del>(</del>	CGB
	480 Temporarily Unavailable ACK	<b>←</b> →	<b>→</b>	CGBA

TP111010	SIP reference: RFC 32	261 [6]		ISUP reference:
T00 (	OID IOLID/Darria and Darriata	f D = = = t = i = = = it ==		27 [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			
OID . I . I'	(GRS) or Circuit group blocking	g message (CG	B) with the indic	ation hardware failure oriented
SIP selection criteria:				
ISUP selection				
criteria:				
Test purpose:	GRS for more than one conne	ctions received,	a BYE is sent for	or each connection
				nding an IAM message for each
	call association on receipt of a		in the confirmed	I state, were the Range and
	Status Parameter value is bigg	er than "1":		
	the SUT shall send a	BYE requests for	or each call asso	ociation.
SIP Parameter				
values:				
ISUP Parameter				
values:				
Comments:	SIP	-	SUT	ISUP
	INVITE(1)	<b>→</b>		IAM
	180 Ringing	<b>←</b>	<b>←</b>	
	200 OK INVITE	<b>←</b>	+	ANM
	ACK	<b>→</b>		
		Conv	ersation	
	INVITE(2)	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
		Conv	ersation	
			+	GRS
	BYE (1)	<b>←</b>	<b>→</b>	GRA
	200 OK BYE	<b>→</b>		
	BYE (2)	<b>←</b>		
	200 OK BYE	<b>→</b>		

TP111011	SIP reference: RFC 32	61 [6]		ISUP reference: 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" 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 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 "1":  • the SUT shall send a BYE requests for each call association.				
SIP Parameter				=	
values:					
ISUP Parameter					
values:					
Comments:	SIP	;	SUT	ISUP	
	INVITE(1)	<b>→</b>	<b>→</b>	IAM	
	180 Ringing	<b>←</b>	+	ACM	
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM	
	ACK	<b>→</b>			
		Conv	ersation		
	INVITE(2)	<b>→</b>	<b>→</b>	IAM	
	180 Ringing	<b>←</b>	+	ACM	
	200 OK INVITE	<b>←</b>	+	ANM	
	ACK	<b>→</b>			
		Conv	ersation		
			<b>←</b>	CGB	
	BYE (1)	<b>←</b>	<b>→</b>	CGBA	
	200 OK BYE	<b>→</b>			
	BYE (2)	<b>←</b>			
	200 OK BYE	<b>→</b>			

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: R	FC 3261 [6]		ISUP reference: 027 [1], clause 7.2.3.2.1.4
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message			
SIP selection	Tool on /Baois sail/soil	ang or the httri in	cooago	
criteria:				
ISUP selection				
criteria:				
Test purpose:	IAM contains the complet of INVITE	te Called party numb	er and the send	ing complete indication, sending
	Ensure that the SUT in idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication:			
OID D	Sends the INVIT	E message.		
SIP Parameter				
values:	IANA Colled martis much			
ISUP Parameter values:	IAM; Called party number	er: with sending com	piete indication	
Comments:	ISUP/BICC	S	SUT	SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM		<del>-</del>	180 Ringing
		Ringi	ng tone	
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conv	ersation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	<b>←</b>	<b>←</b>	200 OK BYE

TP301002	SIP reference: RFC 3261 [6]	ES	ISUP reference: 283 027 [1], clause 7.2.3.2.1.4	
TSS reference:	ISUP-SIP /Basic call/Sending of the INVI	TE 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	mber		
Comments:	ISUP/BICC	SUT	SIP	
	IAM →		→ INVITE	
	ACM		← 180 Ringing	
		Ringing tone		
	ANM ←		← 200 OK INVITE	
			→ ACK	
		Conversation		
	REL →		→ BYE	
	RLC <b>←</b>		← 200 OK BYE	

TP301003	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 a sufficient number of digits to route the call to the called party, sending of INVITE  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.				
SIP Parameter					
values:					
ISUP Parameter values:	IAM; Called party number: complete number				
Comments:	ISUP/BICC S	SUT SIP			
	IAM →	→ INVITE			
	ACM ←	← 180 Ringing			
	Ring	ing tone			
	ANM <b>←</b>	← 200 OK INVITE			
		→ ACK			
		ersation			
	REL →	→ BYE			
	RLC +	★ 200 OK BYE			

TP301004	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:		nding of INVITE  f an IAM message with the minimum number of eceived, by observing the timer Ti/w1 which has		
SIP Parameter				
values:				
ISUP Parameter				
values:				
Comments:	ISUP/BICC S	SUT SIP		
	IAM →			
	$T_I/W$	<sub>1</sub> expiry		
		→ INVITE		
	ACM ←	← 180 Ringing		
	Ringi	ing tone		
	ANM ←	← 200 OK INVITE		
		→ ACK		
	Conv	ersation		
	REL →	→ BYE		
	RLC ←	← 200 OK BYE		

TP301005	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 me	<del></del> :
SIP selection	1001 On / Daoie can/ Containing of the internal internal	secage
criteria:		
ISUP selection	PICS 1/3	
criteria:		
Test purpose:		to "continuity check not required", the INVITE
	is sent immediately	
	Engure that the CLIT is idle state, as receipt of	an IAM massage with the complete collect
	Ensure that the SUT in idle state, on receipt of party number containing the Continuity Chec	
	Indicators parameter is set to indicate "contin	
	linaloatoro parameter lo del te maioate	any encontroquired .
	<ul> <li>sends a INVITE message.</li> </ul>	
SIP Parameter	•	
values:		
ISUP Parameter	IAM: Nature of Connection Indicators paramet	er is set to indicate <b>"continuity check not</b>
values:	required"	
Comments:		SUT SIP
	IAM →	→ INVITE
	ACM	← 180 Ringing
	9	ng tone
	ANM ←	← 200 OK INVITE
	_	→ ACK
		ersation
	REL →	→ BYE
	RLC +	← 200 OK BYE

TP301006	SIP reference: RFC 3261 [6]		ISUP reference: )27 [1], clause 7.2.3.2.1.2	
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE	nessage		
SIP selection criteria:	NOT PICS 4/11			
ISUP selection criteria:	PICS 4/5 AND PICS 1/3			
Test purpose:	IAM received continuity check indicator is set to "continuity check required on this 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 required on this circuit":  • Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful".			
SIP Parameter values:				
ISUP Parameter	IAM: Nature of Connection Indicators param	eter which is set t	o "continuity check required	
values:	on this circuit			
	COT: Continuity Indicators parameter "cont	•		
Comments:	ISUP/BICC	SUT	SIP	
	IAM →	_		
	COT(successful)	<b>→</b>	INVITE	
	ACM ←	<b>←</b>	180 Ringing	
		ging tone	000 014 10 11 17 17 17 17 17 17 17 17 17 17 17 17	
	ANM ←	<del>(</del>	200 OK INVITE	
		<b>→</b>	ACK	
		versation	DVE	
	REL →	<b>→</b>	BYE	
	RLC ←		200 OK BYE	

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 circuit", the INVITE is sent after COT "success Ensure that the SUT in idle state, on receipt of party number containing the Continuity Check	an IAM message with the complete called						
	Indicators parameter which is set to "continui	ty check performed on previous circuit": the Continuity message with the Continuity						
SIP Parameter values:								
ISUP Parameter	IAM Nature of Connection Indicators parameter	er which is set to "continuity check performed						
values:	on previous circuit"							
	COT: Continuity Indicators parameter "contin	uity check successful"						
Comments:	ISUP/BICC S	SUT SIP						
	IAM →							
	COT(successful) →	→ INVITE						
	ACM ←	← 180 Ringing						
	Ringi	ng tone						
	ANM ←	◆ 200 OK INVITE						
		→ ACK						
		ersation						
	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	PICS 4/5 AND PICS 4/11						
criteria:							
ISUP selection	PICS 1/3 AND PICS 4/2						
criteria: Test purpose:	IAM continuity check required received, prec	andition request in the INV/ITE					
rest purpose.	IAM Continuity Check required received, prec	onalion request in the INVITE					
	Ensure that the SUT in idle state, on receipt indicator in the Nature of Connection Indicator "continuity check required on this circuit"	•					
	SDP offer or answer carrying the co when the Continuity message with t	condition using the SDP offer in the INVITE. The infirmation of a precondition being met is sent he Continuity Indicators parameter set to s received and the requested preconditions are					
SIP Parameter	INVITE: Require: precondition						
values:	SDP a=curr:qos local none						
	a=curr:qos remote none a=des:qos mandatory local se	odrecy					
	a=des:qos mandatory local sel						
	183: Require: 100rel						
	SDP a=curr:qos local none						
	a=curr:qos remote none	odroov.					
	a=des:qos mandatory local sei a=des:qos mandatory remote :						
	a=conf:gos remote sendrecv	351141007					
	·						
	UPDATE:						
	SDP a=curr:qos local sendrecv a=curr:qos remote none						
	a=des:qos mandatory local se	ndrecv					
	a=des:qos mandatory remote						
	200 OK LIDDATE						
	200 OK UPDATE SDP a=curr:qos local sendrecv						
	a=curr:gos remote sendrecv						
	a=des:qos mandatory local se	ndrecv					
	a=des:qos mandatory remote						
ISUP Parameter values:	IAM: Nature of Connection Indicators parameter on this circuit	eter which is set to "continuity check required					
values.	COT: Continuity Indicators parameter "conti	nuity check successful"					
Comments:	ISUP/BICC	SUT SIP					
	IAM →	→ INVITE					
		← 183 Session Progress					
		→ PRACK					
		← 200 OK PRACK					
	COT →	→ UPDATE ← 200 OK UPDATE					
	ACM ←	€ 180 Ringing					
	AOW	→ PRACK					
		€ 200 OK PRACK					
	Rin	ging tone					
	ANM ←	← 200 OK INVITE					
		→ ACK					
		versation					
	REL →	→ BYE					
	RLC ←	← 200 OK BYE					

TP301009	SIP reference: RFC 3261 [6]	ES 202 027	ISUP reference:				
T00 (	10115 015 5 110 15 (11 15)	· · · · · · · · · · · · · · · · · · ·	[1], clause 7.2.3.2.1.2, 7.2.3.2.3				
TSS reference: SIP selection	ISUP-SIP/Basic call/Sending of the INVITE message						
criteria:	PICS 4/5 AND PICS 4/11						
ISUP selection criteria:	PICS 1/3 AND PICS 4/2						
Test purpose:	IAM continuity check performed on a pre INVITE	evious circuit receive	d, precondition request in the				
	Ensure that the SUT in idle state, on recindicator in the Nature of Connection Incontinuity check performed on previous	licators parameter in					
	SDP offer or answer carrying the when the Continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity message with the continuity message was a support of the continuity was a support of the continui	ne confirmation of a posith the Continuity Inc					
SIP Parameter values:	INVITE: Require: precondition  SDP a=curr:qos local none a=curr:qos remote none a=des:qos mandatory loca a=des:qos none remote s						
	183: Require: 100rel SDP						
	UPDATE: SDP a=curr:qos local sendrecv a=curr:qos remote none a=des:qos mandatory loca a=des:qos mandatory rem	al sendrecv					
	200 OK UPDATE  SDP						
ISUP Parameter values:	IAM Nature of Connection Indicators par on previous circuit" COT: Continuity Indicators parameter "c		•				
Comments:	ISUP/BICC	SUT	SIP				
	IAM →	→					
		<del>(</del>					
		<b>→</b>	PRACK				
	_	<del>(</del>					
	COT →	→ ←	*				
	ACM ←	÷	180 Ringing				
		<del>(</del>					
	ANM ←	Ringing tone					
		Conversation	ACK				
	REL →	Conversation -	BYE				
	RLC ←	<del>-</del>					
-							

TP301010	SIP reference: RFC 3261 [6]		ISUP reference: )27 [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 (INIII) Ensure that if no calling party number is received an INR message to request the calling until receiving an INF message with calling pareceived in the INF message, O-MGCF may reconfiguration.	ved in the incom party number an rty number. If no	nd not sends the INVITE request o calling party number is					
SIP Parameter								
values:								
ISUP Parameter								
values:								
Comments:	ISUP/BICC S	SUT	SIP					
	IAM →							
	INR <b>←</b>							
	INF →							
		<b>→</b>	INVITE					
	ACM ←	<b>←</b>	180 Ringing					
	Ringing tone							
	ANM ←	<b>←</b> →	200 OK INVITE ACK					
	Conversation							
	REL →	<b>→</b>	BYE					
	RLC ←	+	200 OK BYE					

TP301011	SIP reference: RFC 3261	[6]	ES 283	ISUP reference: 027 [1], clause 7.2.3.2.1a
TSS reference:	ISUP-SIP/Basic call/Sending of the	e INVITE me	essage	
SIP selection criteria:				
ISUP				
selection				
criteria:				
Test purpose:	Sending of INVITE without detern		_	·
	Ensure that if the O-MGCF sends determined, the O-MGCF shall: - start timer Ti/w2; and - be prepared to process SAM - be prepared to handle incomir		·	ne end of address signalling is
	On receipt of a SAM from the ISU stop timer Ti/w3 (if it is running); send an INVITE request complyin	P side, the 0	O-MGCF shall:	for this call in the Request-URI.
SIP	1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1			
Parameter				
values:				
ISUP Parameter values:				
Comments:	ISUP/BICC		SUT	SIP
	IAM ·	<b>→</b>	_	
			<b>→</b>	INVITE 404/484
			<b>→</b>	ACK
	SAM	<b>→</b>	•	AOR
			<b>→</b>	INVITE
			<b>←</b>	10 1/ 10 1
			<b>→</b>	ACK
	SAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
		•	ng tone	
	ANM	←	<b>←</b> →	200 OK INVITE ACK
			ersation	
	1	<del>)</del>	<b>→</b>	BYE
	RLC	<del>(</del>	<b>+</b>	200 OK BYE

TP301012	SIP reference: RFC 3	3261 [6]		ISUP reference:			
			ES 283 (	27 [1], clause 7.2.3.2.2.2			
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message						
SIP selection	Based on table 8						
criteria:							
ISUP selection							
criteria:							
Test purpose:	Mapping of TMR into the SDF	oin the sent INV	ITE				
				ssage, with the <b>Transmission</b>			
	Medium Requirement (TMR)	) parameter set t	o IMR_VALU	E:			
				escription defined with the "a="			
OID D	"b=" and "m=" lines s		NE_VALUE.				
SIP Parameter	INVITE: a_b_m_LINE_VALU	=					
values:							
ISUP Parameter	IAM: TMR: ISUP_TMR						
values:				OID			
Comments:	ISUP/BICC	SU		SIP			
	IAM	<b>→</b>	<b>→</b>	INVITE			
	ACM	<b>←</b>	<b>←</b>	180 Ringing			
		Ringin	g tone				
	ANM	<b>←</b>	+	200 OK INVITE			
			<b>→</b>	ACK			
		Conver	sation				
	REL	<b>→</b>	<b>→</b>	BYE			
	RLC	<del>(</del>	+	200 OK BYE			

TP301013	SIP reference: RFC 3261 [6]			ISUP reference:
			ES 283 0	27 [1], clause 7.2.3.2.2.2
TSS reference:	ISUP-SIP/Basic call/ Sending of the IN	NVITE me	ssage	
SIP selection	Based on table 9			
criteria:				
ISUP selection				
criteria:				
Test purpose:	Mapping of USI into the SDP in the se	ent INVITE	Ī	
	Ensure that the SUT in the idle state of		of an IAM me	essage, with the <b>user</b>
	information parameter set to USI_V	ALUE:		
				ion defined with the "a = " "b="
CID Donomotor	and "m=" lines set to a_b_m	_LINE_V	ALUE.	
SIP Parameter values:	INVITE: a_b_m_LINE_VALUE			
ISUP Parameter	IAM: LICI, ICUD LICI			
values:	IAM: USI: ISUP_USI			
Comments:	ISUP/BICC	SUT	•	SIP
Comments.	IAM →	301	→	INVITE
	ACM ←		<del>-</del>	
	ACIVI	Dinging	=	180 Ringing
	ANM ←	Ringing		200 OK INIVITE
	ANM ←		<del>(</del>	200 OK INVITE
		0	<b>→</b>	ACK
	DE!	Convers		D)/E
	REL →		<b>→</b>	BYE
	RLC ←		<del>-</del>	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 receicall, with the user information parameter see sends an INVITE message, with the	et to USI_VALUE:						
SIP Parameter	Offer: m=audio RTP/AVP dynamic PT							
values:	a = rtpmap dynamic PT AMR							
	Answer: m=audio RTP/AVP dynamic PT a = rtpmap dynamic PT AMR							
ISUP Parameter	IAM: USI= USI_VALUE (PIXIT)							
values:								
Comments:	ISUP/BICC S	UT SIP						
	IAM →	→ INVITE						
	ACM ←	<ul><li>180 Ringing</li></ul>						
	Ringir	ng tone						
	ANM <b>←</b>	◆ 200 OK INVITE						
		→ ACK						
		ersation						
	REL →	→ BYE						
	RLC ←	← 200 OK BYE						

TP301015	SIP reference:	RFC 3261 [6]		ISUP reference:
			ES 283 (	)27 [1], clause 7.2.3.2.2.2
TSS reference:	ISUP-SIP/Basic call/ Se	ending of the INVITE m	essage	
SIP selection	NOT PICS 2/8			
criteria:				
ISUP selection				
criteria:				
Test purpose:	No AMR codec in the S	DP, when no equipme	nt implements	the AMR codec
	Ensure that the SUT in	the Idle state on receip	ot of an IAM m	essage indicating a speech
				and the IMS network serves
	that no user equipment	implements the AMR of	codec, then the	e AMR codec shall be
	excluded from the SDP	offer.		
SIP Parameter	INVITE: SDP no AMR of	codec		
values:				
ISUP Parameter	IAM: USI= USI_VALUE	(PIXIT)		
values:				
Comments:	ISUP/BICC	SU	T	SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
		Ringing	tone	
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conver	sation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	É	ŕ	200 OK BYE
	INLO			ZOO ON DIL

Table 8

	Values for test purposes TP301012								
	ISUP SDP - a_b_m_LINE_VALUE								
	TMR parameter		m= li	ine	b= line	a= line			
	TMR codes	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>: <bandwidt h-value&gt;</bandwidt </modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters&gt;</clock></encoding></dynamic-pt>			
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)</dynamic-pt></dynamic-pt>			
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</dynamic-pt>			
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</dynamic-pt>			

Table 9

	1			ues for test pur	rposes TP3	801012, TP30			
VA		ISU				SDP - a_b_m_LINE_VALUE			
		USI para	meter	HLC IE in ATP	m= line b= line			a= line	
	TMR	Information Transport Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristi cs Identification	<media></media>	<transport< th=""><th><fmt-list></fmt-list></th><th><modifier>: <bandwidth -value&gt;</bandwidth </modifier></th><th>rtpmap:<dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters&gt;</clock></encoding></dynamic-pt></th></transport<>	<fmt-list></fmt-list>	<modifier>: <bandwidth -value&gt;</bandwidth </modifier>	rtpmap: <dynamic-pt> <encoding name="">/<clock rate="">[/encoding parameters&gt;</clock></encoding></dynamic-pt>
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&gt; PCMA/8000) (see note 1)</dynamic- </dynamic-pt>
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</dynamic-pt>
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)</dynamic-pt>

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.

TP301016	SIP reference	: RFC 3261 [6]		ISUP reference:				
			ES 283	3 027 [1], clause 7.2.3.2.2.2				
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message							
SIP selection	PICS 2/8							
criteria:								
ISUP selection								
criteria:								
Test purpose:	The SUT defines the I	RS and RR bandwidth n	nodifiers whe	en AMR codec is used				
				call is incoming, the O-MGCF				
			P RR and R	S bandwidth modifiers specified				
OID D	in RFC 3556 [17] to di							
SIP Parameter	INVITE: SDP b=RS:<							
values:	b=RR:<	bandwidth-value>						
ISUP Parameter								
values:								
Comments:	ISUP/BICC	SUT	-	SIP				
	IAM	<b>→</b>	<b>→</b>	INVITE				
	ACM	<del>(</del>	<b>←</b>	180 Ringing				
		Ringin	g tone					
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE				
			<b>→</b>	ACK				
		Conve	rsation					
	REL	<b>→</b>	<b>→</b>	BYE				
	RLC	<del>-</del>	+	200 OK BYE				

TP301017	SIP reference: RFC 3261	[6]		ISUP reference: 027 [1], clause 7.2.3.2.2.1			
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message						
SIP selection	l l l l l l l l l l l l l l l l l l l		.ooougo				
criteria:							
ISUP selection criteria:							
Test purpose:	Mapping of Called party number into the To header user=phone is included						
	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 <b>To header field</b> which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.						
SIP Parameter	INVITE: To: sip:; user=phone						
values:							
ISUP Parameter values:							
Comments:	ISUP/BICC	SU	IT	SIP			
	IAM →		<b>→</b>	INVITE			
	ACM ←		<b>←</b>	180 Ringing			
		Ringin	g tone				
	ANM ←		<b>+</b>	200 OK INVITE			
			<b>→</b>	ACK			
		Conve	rsation				
	REL →		<b>→</b>	BYE			
	RLC ←		+	200 OK BYE			

TP301018	SIP reference: RFC 3261 [6]			ISUP reference: 027 [1], clause 7.2.3.2.2.1				
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message							
SIP selection			3-					
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 <b>To header field</b> .							
SIP Parameter	INVITE: To:							
values:								
ISUP Parameter								
values:								
Comments:	ISUP/BICC	SUT		SIP				
	IAM →		<b>→</b>	INVITE				
	ACM ←		<b>←</b>	180 Ringing				
		Ringing ton						
	ANM ←		<b>←</b>	200 OK INVITE				
			<b>→</b>	ACK				
		Conversatio						
	REL →		<b>→</b>	BYE				
	RLC +		+	200 OK BYE				

TP301019	SIP reference: RFC	3261 [6]			ISUP reference: 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 <b>To header field</b> which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.							
SIP Parameter values:	INVITE: To: sip:; user=p	phone						
ISUP Parameter values:								
Comments:	ISUP/BICC		SUT		SIP			
	IAM	<b>→</b>		<b>→</b>	INVITE			
				<b>←</b>	404/484			
				<b>→</b>	ACK			
	SAM	<b>→</b>		<b>→</b>	INVITE			
	ACM	<b>←</b>	Ringing tone	<b>←</b>	180 Ringing			
	ANM	<b>←</b>		<b>←</b> →	200 OK INVITE ACK			
			Conversation	7	AUN			
	REL	<b>→</b>	Conversation	<b>→</b>	BYE			
	RLC	<del>7</del> ←		<del>7</del>	200 OK BYE			
	INLU				ZUU UN DIE			

TP301020	SIP reference: F	RFC 3261 [6]	ES 283 (	ISUP reference: 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:	shall use the Hop Count different default values ( Forwards header and th the Hop Counter to the I Max-Forwards for a give	counter procedure is so ter parameter to derive that are based on net e Hop Counter, an ad Max Forwards at the Cen message should be	upported in the the the Max-Forwork demands aptation mechorMGCF.	e CS network, the O-MGCF wards SIP header. Due to the s/provisions) of the SIP Max- anism shall be used to adopt creasing with each successive nd similarly for Hop Counter.
SIP Parameter values:				
ISUP Parameter values:				
Comments:	ISUP/BICC	SU	IT	SIP
	IAM	<b>→</b>	→	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
		Ringin	g tone	
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conver	sation	
	REL	<b>→</b>	→	BYE
	RLC	<b>←</b>	+	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 w	vith the used (imp	olemented) 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 paramer Request URI and To header of the INVITE Ruser=phone" it shall contain an Internationa a "+" sign (e.g. tel:+4911231234567). If the Request URI is a sip URI with "user=platelecommunication number prefixed by a "+" sip:+4911231234567@host). Ensure that the SUT is mapping the Called For Called Party Number parameter, Nature of a IAM:	ter of the IAM message is used to derive equest. If the Request URI is a tel URI with I public telecommunication number prefixed by none" it shall contain an International public sign and a host portion (e.g.
SIP Parameter	INVITE: To:, Request URI	
values: ISUP Parameter		
values:		
Comments:	ISUP/BICC S	UT SIP
	IAM →	→ INVITE
	ACM ←	← 180 Ringing
	Ringi	ng tone
	ANM ←	← 200 OK INVITE
		→ ACK
		ersation
	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 r		
SIP selection	3 · · ·		
criteria:			
ISUP selection	NOT PICS 1/6		
criteria:			
Test purpose:	Mapping of Called party number "national (significant) number" into the To header and Request URI user=phone is included		
	Address signal; the format of the To header field is "	quest URI is a tel URI with "user=phone" and nunication number prefixed by a "+" sign (e.g. one" it shall contain an International public sign and a host portion (e.g. arty address information contained in the ddress = "National (significant) number" and Request URI inserting "+" CC before the	
SIP Parameter	INVITE: To:		
values: ISUP Parameter			
values:			
Comments:	ISUP/BICC SI	JT SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	Ringir	ig tone	
	ANM <b>←</b>	← 200 OK INVITE	
		→ ACK	
		rsation	
	REL →	→ BYE	
	RLC <b>←</b>	← 200 OK BYE	

SIP reference: RFC 326 <sup>2</sup>	1 [6]		ISUP reference:
			)27 [1], clause 7.2.3.2.2.7
ISUP-SIP/Basic call/ Sending of t	the INVITE m	essage	
PICS 4/18			
Mapping of USI parameter into P	STN XML Be	earerCapability	,
1	•	f an IAM mess	age <b>User Service</b>
Information (USI) set to USI_VA	LUE:		
	age with the	PSTN XML <b>B</b> e	earer Capability (BC) set to
_			
INVITE; PSTN XML BearerCapa	i <b>bility</b> (BC): L	JSI_VALUE (F	TIXIY)
IAM; <b>USI:</b> USI_VALUE (PIXIT)			
ISUP/BICC	SU	JT	SIP
IAM =	•	<b>→</b>	INVITE
ACM	-	<b>←</b>	180 Ringing
	Ringing	g tone	
ANM		<b>+</b>	200 OK INVITE
		<b>→</b>	ACK
	Conver	sation	
REL		<b>→</b>	BYE
		<b>É</b>	200 OK BYE
	ISUP-SIP/Basic call/ Sending of the PICS 4/18  Mapping of USI parameter into Pics that the SUT in Idle state Information (USI) set to USI_VALUE.  • sends the INVITE mess USI_VALUE.  INVITE; PSTN XML BearerCapa  IAM; USI: USI_VALUE (PIXIT)  ISUP/BICC IAM ACM ANM REL	Mapping of USI parameter into PSTN XML Beter Ensure that the SUT in Idle state, on receipt of Information (USI) set to USI_VALUE:  • sends the INVITE message with the USI_VALUE.  INVITE; PSTN XML BearerCapability (BC): USI_VALUE (PIXIT)  ISUP/BICC SUSI_VALUE (PIXIT)	ISUP-SIP/Basic call/ Sending of the INVITE message

TP301024	SIP reference: RFC 3261 [6]	ISUP reference:	
<i>(</i>	10115 015/5 : 11/0 !: (4 15/6/175	ES 283 027 [1], clause 7.2.3.2.2.7	
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE m	nessage	
SIP selection	PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of USI and USI prime parameter into		
	Ensure that the SUT in Idle state, on receipt o		
	the <b>USI Prime</b> set to "unrestricted digital infor	mation with tones and announcements":	
		first PSTN XML BearerCapability set to	
		cond PSTN XML BearerCapability set to	
	•	tones and announcements" (the USI prime	
010.0	value).		
SIP Parameter	INVITE; first PSTN XML Bearer Capability: speech		
values:	second PSTN XML Bearer Capability	y: unrestricted digital information with	
	tones and announcements		
ISUP Parameter	IAM; USI: speech		
values:	USI Prime: unrestricted digital information wit		
Comments:	ISUP/BICC SU		
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	Ringin	g tone	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Conve	rsation	
	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 m		
SIP selection criteria:	PICS 4/18		
ISUP selection criteria:			
Test purpose:	Mapping of FCI "ISUP not used all the way" in Ensure that the SUT in Idle state, on receipt of the state of t		
		FN XML ProgressIndicator set to "call is not as information is available in-band (#1)".	
SIP Parameter	INVITE;		
values:	PSTN XML ProgressIndicator: call is not end-to-end ISDN: further call progress information is available in-band (#1)		
ISUP Parameter	IAM; ISUP indicator: ISUP not used all the w	ay	
values:			
Comments:	ISUP/BICC SU	JT SIP	
	IAM →	→ INVITE	
	ACM ←	<ul> <li>180 Ringing</li> </ul>	
	Ringin	g tone	
	ANM ←	← 200 OK INVITE	
		→ ACK	
	Conve	rsation	
	REL →	→ BYE	
	RLC ←	← 200 OK BYE	

TP301026	SIP reference: RFC 3261 [6]		ISUP reference:
		ES 283 (	)27 [1], clause 7.2.3.2.2.8
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE	message	
SIP selection	PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of FCI "Originating access non ISD	N" into PSTN X	ML ProgressIndicator #3
	Forever that the OUT in July at the comments	-6 1004	and a section of the LOUID
	Ensure that the SUT in Idle state, on receipt		
	indicator set to "ISUP used all the way" and access non-ISDN":	the ISDN acce	ss indicator set to originating
	access non-isbly .		
	sends the INVITE message with the	DSTN YMI D	ograssindicator sa tot
	"Originating access is non ISDN (#3		ogressificator se tot
SIP Parameter	INVITE:	·) ·	
values:	PSTN XML ProgressIndicator: Originating access is non ISDN (#3)		
ISUP Parameter	IAM:		
values:	ISUP indicator: ISUP used all the way		
	ISDN access indicator: originating access r	on-ISDN	
Comments:		UT	SIP
	IAM →	<b>→</b>	INVITE
	ACM <b>←</b>	<b>←</b>	180 Ringing
	Ringi	ng tone	5 5
	ANM	<b>←</b>	200 OK INVITE
		<b>→</b>	ACK
	Conve	ersation	
	REL →	<b>→</b>	BYE
	RLC <del>C</del>	<b>←</b>	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 n	:	
SIP selection criteria:	PICS 4/18		
ISUP selection criteria:			
Test purpose:	Mapping of FCI "ISUP used all the way" and XML ProgressIndicator #3  Ensure that the SUT in Idle state, on receipt of indicator, set to "ISUP used all the way" and		
	access non-ISDN":	the IODIN access indicator set to originating	
		PSTN XML ProgressIndicatorPSTN XML	
	ProgressIndicator set to "Originating		
SIP Parameter	INVITE; PSTN XML ProgressIndicatorPSTN XML ProgressIndicator: Originating		
values:	access id non ISDN (#3)		
ISUP Parameter	IAM; ISUP indicator: ISUP used all the way		
values:	ISDN access indicator: originating access ne	on-ISDN	
Comments:	ISUP/BICC SI	JT SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	Ringin	ig tone	
	ANM ←	€ 200 OK INVITE	
		→ ACK	
	Conve	rsation	
	REL →	→ BYE	
	RLC ←	← 200 OK BYE	

TP301028	SIP reference: RFC 3261 [6]		ISUP reference:
			027 [1], clause 7.2.3.2.2.8
TSS reference:	ISUP-SIP/Basic call/ Sending of the INV	ITE message	
SIP selection	PICS 4/18		
criteria:			
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 acco	ess indicator set to "originating
	access ISDN":		
	<ul> <li>sends the INVITE message with "originating access ISDN" (#6).</li> </ul>	n the PSTN XML P	rogressIndicator set to
SIP Parameter	INVITE; PSTN XML ProgressIndicator: "originating access ISDN" (#6)		
values:			
ISUP Parameter	IAM; ISUP indicator: ISUP used all the way		
values:	ISDN access indicator: originating acce	ess ISDN	
Comments:	ISUP/BICC	SUT	SIP
	IAM →	<b>→</b>	INVITE
	ACM ←	<b>←</b>	180 Ringing
	R	linging tone	
	ANM ←	<b>←</b>	200 OK INVITE
		<b>→</b>	ACK
	C	onversation	
	REL →	<b>→</b>	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:  SIP Parameter values:	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.  INVITE; PSTN XML ProgressIndicator: "originating access ISDN" (#6) and PSTN XML ProgressIndicator: PI_VALUE (PIXIT)		
ISUP Parameter	IAM; ISUP indicator: ISUP used all the way		
values:	ISDN access indicator: originating access IS ATP progress indicator: PI_VALUE (PIXIT)	SDN	
Comments:	ISUP/BICC SU	JT SIP	
	IAM →	→ INVITE	
	ACM ←	<ul> <li>180 Ringing</li> </ul>	
	Ringin	•	
	ANM <b>←</b>	← 200 OK INVITE	
		→ ACK	
	Conver		
	REL →	→ BYE	
	RLC <del>C</del>	★ 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	PICS 4/18	····osaago	
criteria:			
ISUP selection			
criteria:			
Test purpose:	Mapping of ATP contains a LLC into a PSTN	NXML LLC in the sent INVITE	
	Ensure that the SUT in Idle state, on receipt Transport Parameter (ATP) containing the LLLC_VALUE:  • sends the INVITE message with the LLC_VALUE.		
SIP Parameter values:	the PSTN XML LowLayerCompatibility: LLC_VALUE (PIXIT)		
ISUP Parameter	IAM; ATP LLC: LLC_VALUE (PIXIT)		
values:			
Comments:		SUT SIP	
	IAM →	→ INVITE	
	ACM <b>←</b>	← 180 Ringing	
		ng tone	
	ANM ←	← 200 OK INVITE	
		→ ACK	
		ersation	
	REL →	→ BYE	
	RLC +	← 200 OK BYE	

TP301031	SIP reference: RFC 3261 [6]			ISUP reference:
			ES 283 (	)27 [1], clause 7.2.3.2.2.8
TSS reference:	ISUP-SIP/Basic call/ Sending of the I	NVITE m	essage	
SIP selection	PICS 4/18			
criteria:				
ISUP selection				
criteria:				
Test purpose:	Mapping of ATP contains a HLC into	a PSTN	XML HLC in tl	ne sent INVITE
	Ensure that the SUT in Idle state, on			
	Transport Parameter (ATP) containin	g the <b>Hiç</b>	gh Layer Com	patibility (HLC) set to
	HLC_VALUE:			
		with the	PSTN XML Hi	ghLayerCompatibility set to
	HLC_VALUE.			(5))((5)
SIP Parameter	INVITE: PSTN XML HighLayerCom	oatibility	: HLC_VALUE	(PIXIT)
values:				
ISUP Parameter	IAM; ATP HLC: HLC_VALUE (PIXIT)			
values:			_	
Comments:	ISUP/BICC	SU		SIP
	IAM →		<b>→</b>	INVITE
	ACM <b>←</b>		+	180 Ringing
		Ringing	g tone	
	ANM ←		<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conver	rsation	
	REL →		<b>→</b>	BYE
	RLC ←		+	200 OK BYE

TP301032	SIP reference: RFC 3261 [6]	ISUP reference:
11 30 1032	on reference. At 6 0201 [0]	ES 283 027 [1], clause 7.2.3.2.2.8
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE	
SIP selection	PICS 4/18	moodago
criteria:	1100 1/10	
ISUP selection		
criteria:		
Test purpose:	Mapping of User Teleservice Information par INVITE	rameter into a PSTN XML HLC in the sent
	Ensure that the SUT in Idle state, on receipt Teleservice containing the <b>High Layer Com</b>	patibility (HLC) set to HLC_VALUE:
	<ul> <li>sends the INVITE message with the HLC_VALUE.</li> </ul>	e PSTN XML HighLayerCompatibility set to
SIP Parameter	INVITE: PSTN XML HighLayerCompatibili	y: HLC_VALUE (PIXIT)
values:		(ALLIE (DI)(IT)
ISUP Parameter values:	IAM; User Teleservice Information: HLC_\	ALUE (PIXII)
Comments:	ISUP/BICC S	UT SIP
	IAM →	→ INVITE
	ACM ←	← 180 Ringing
	Ringi	ng tone
	ANM ←	← 200 OK INVITE
		→ ACK
	Conv	ersation
	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	PICS 4/18	-
criteria:		
ISUP selection		
criteria:		
Test purpose:	Mapping of two HLC parameter contained in HighLayerCompatibility elements  Ensure that the SUT in Idle state, on receipt	t of an IAM message containing the Access
	Transport Parameter (ATP) containing two I respectively HLC_VALUE1 and HLC_VALU  • sends the INVITE message with two same order HLC_VALUE1 and HLC	vo PSTN XML HighLayerCompatibility in the
SIP Parameter values:	INVITE; first PSTN XML HighLayerCompatibility: second PSTN XML HighLayerCompatibili	
ISUP Parameter	IAM;	
values:	ATP first HLC: HLC_VALUE1 (PIXIT) ATP second HLC: HLC_VALUE2 (PIXIT)	
Comments:	ISUP/BICC S	SUT SIP
	IAM →	→ INVITE
	ACM ←	<ul> <li>180 Ringing</li> </ul>
	Ring	ing tone
	ANM ←	← 200 OK INVITE
		→ ACK
		versation
	REL →	→ BYE
	RLC <b>←</b>	← 200 OK BYE

TP301034	SIP reference: RF	C 3261 [6]		ISUP reference:
			ES 283 (	)27 [1], clause 7.2.3.2.2.8
TSS reference:	ISUP-SIP/Basic call/ Send	ing of the INVITE m	essage	
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	Mapping of calling party ca	ategory into cpc par	ameter in the l	P-Aserted-Identity
	Ensure that the SUT map	• • •	• •	•
	l.	ed-Identity and Acce	ept-Contact he	ader parameter "language"
	SIP_LANG.			
SIP Parameter	INVITE; P-Asserted-Identif	ty, Accept-Contact		
values:				
ISUP Parameter	IAM; Calling party category	У		
values:				
Comments:	ISUP/BICC	SU	JT	SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	+	180 Ringing
		Ringin	g tone	
	ANM	<b>←</b>	<b>+</b>	200 OK INVITE
			<b>→</b>	ACK
		Conve	sation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	<b>←</b>	<b>←</b>	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	ce: RFC 3261 [6]	ISUP reference:
			ES 283 027 [1], clause 7.2.3.2.1.4
TSS reference:	ISUP-SIP/Basic call/F	Receipt of SAM after INVIT	TE has been sent
SIP selection	PICS 3/1		
criteria:			
ISUP selection	PICS 3/5 AND NOT I	PICS 3/8	
criteria:			
Test purpose:	Overlap procedure no	ot supported, SAM is ignor	red
			sing towards the SIP network, subsequent
	SAMs received after	the SUT has sent the INVI	ITE are ignored.
SIP Parameter			
values:			
ISUP Parameter	SAM; subsequent n	umber (PIXIT)	
values:			
Comments:	ISUP/BICC	SUT	SIP
	IAM	<b>→</b>	→ INVITE
	SAM	<b>→</b>	
	ACM	<b>←</b>	← 180 Ringing
	Ring	ging tone	
	ANM	<b>←</b>	← 200 OK INVITE
			→ ACK
		Convers	sation
		<b>→</b>	→ BYE
	RLC	<del>(</del>	← 200 OK BYE

TP302002	SIP reference: RFC	2261 [6]		ISUP reference:
17302002	Sir reference. Kro	. 3201 [0]	ES 2	283 027 [1], clauses 7.2.3.2.1a and 7.2.3.2.1.4
TSS reference:	ISUP-SIP/Basic call/Receip	t of SAM after invite	has be	-
SIP selection criteria:	PICS 3/2	t or or twi ditor invite	71140 50	on som
ISUP selection criteria:	PICS 3/8			
Test purpose:	Overlap procedure supporte indication received	ed by determining tl	he end d	of address signalling. sending complete
	Check indicator in the Natur	re of Connection Inc uired" on receipt of	dicators a SAM	message containing the Continuity parameter which is set to indicate containing the complete called party SUT
	On receipt of a SAM from the	e ISUP the SUT sh	nall:	
	Stop timer TOIW3 (if it is run TOIW2 shall be restarted an a) The Request-URI received so far for	nd the SUT shall invand the To header		following procedures: he new INVITE shall contain all digits
	b) A new INVITE with INVITE is sent.	the same Call-ID a	and Froi	m header (including tag) as the previous
	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 →		<b>→</b>	INVITE
			<del>-</del>	404/484
			<b>→</b>	ACK
	SAM -		<b>→</b>	INVITE
	5,		<b>+</b>	404/484
			<b>→</b>	ACK
	SAM →		<b>→</b>	INVITE
			<b>←</b>	404/484
			<b>→</b>	ACK
	SAM(F) →		<b>→</b>	INVITE
	ACM		<b>←</b>	180 Ringing
	Ringing to	ne		5 5
	ANM		<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conve	rsation	
	<b>→</b>		<b>→</b>	BYE
	RLC ←	·	<b>←</b>	200 OK BYE

TP302003	SIP re	ference: RFC 3261	[6]		ISUP reference:
				ES 2	283 027 [1], clauses 7.2.3.2.1a and 7.2.3.2.1.4
TSS reference:	ISUP-SIP/Bas	sic call/Receipt of SAI	M after invit	e has be	
SIP selection criteria:	PICS 3/2				
ISUP selection criteria:	PICS 3/8				
Test purpose:	Ensure that the indicator in the check not remaitional number sends an INV  Stop timer TOTOIW2 shall to a The Remaitional number section in the received by A new INVITE c) The new resour reserve param	in the national numbers of the SUT in Idle state, of the Nature of Connection of the Nature of Connection of the Idle of the I	on receipt of on Indicators a SAM and ched, the SU ing all digits SUT shall in to header field a Call-ID are ain a new SU been reserver reflected warm on the control of th	re reach an IAM s parame the max JT s receive voke the eld of the od From DP offer rithin the	of address signalling. Maximum number ned  message containing Continuity Check eter which is set to indicate "continuity kimum number of digits used in the ed in the IAM and the SAM(s).  e following procedures: e new INVITE shall contain all digits header (including tag) as the previous in the O-MGCF may re-use any this call. This re-use of existing is precondition attributes for the SDP orked from the parameters of the
SIP Parameter values:					
ISUP Parameter values:					
Comments:	ISUP/BICC		SUT		SIP
	IAM	<b>→</b>		<b>→</b>	INVITE
				<b>←</b>	404/484
				<b>→</b>	ACK
	SAM	<b>→</b>		<b>→</b>	INVITE
	SAIVI	7		<del>-</del>	404/484
				7	404/464 ACK
				7	ACK
	SAM	<b>→</b>		<b>→</b>	INVITE
	SAIVI	7		<del>-</del>	404/484
				<b>→</b>	ACK
				7	ACK
	SAM	<b>→</b>		<b>→</b>	INVITE
	ACM	<b>←</b> Ringing tone		<del>(</del>	180 Ringing
	ANM	tunging tone		<b>←</b>	200 OK INVITE
		•		<b>→</b>	ACK
			Conve	ersation	//OIX
		<b>→</b>	CONTR	=rsation	BYE
	RLC	<b>+</b>		É	200 OK BYE
	INLO	1			200 ON DIL

TP302004	SIP ref	erence: RFC 3261 [6	61		ISUP reference:
55200-		5. 5. 1. 5 0 20 1 [0		ES 2	283 027 [1], clauses 7.2.3.2.1a and 7.2.3.2.1.4
TSS reference:	ISUP-SIP/Basi	c call/Receipt of SAM	after invite	has be	en sent
SIP selection criteria:	PICS 3/2				
ISUP selection criteria:	PICS 3/8				
Test purpose:		dure supported by det een received to route			of address signalling. Sufficient number
	indicator in the check not req	Nature of Connection	n Indicators SAM and t	parame	message containing Continuity Check eter which is set to indicate "continuity icient number of digits has been
	sends an INVIT	TE message containin	ng all digits	receive	d 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.				
	<ul> <li>b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent.</li> <li>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.</li> <li>d) All other contents of the new INVITE are interworked from the parameters of the original IAM.</li> </ul>				
SIP Parameter	originar	ii dvi.			
values: ISUP Parameter					
values:					
Comments:	ISUP/BICC		SUT		SIP
Comments.	IAM	<b>→</b>	001	→	INVITE
	II UVI	•		÷	404/484
				<b>→</b>	ACK
	SAM	<b>→</b>		<b>→</b>	INVITE
	SAIVI	•		<b>+</b>	404/484
				<b>→</b>	
	SAM	<b>→</b>		<b>→</b>	INVITE
				<b>←</b>	404/484
				<b>→</b>	ACK
	SAM	<b>→</b>		<b>→</b>	INVITE
	ACM	Einging tone		<b>←</b>	180 Ringing
	ANINA	Ringing tone		_	200 OK INIVITE
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE
			0		ACK
			Conve	rsation	DVE
	DI C	<b>→</b>		<b>→</b>	BYE
	RLC	+		<del>-</del>	200 OK BYE

TP302004	SIP reference: RFC 3	3261 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.2.4
TSS reference:	ISUP-SIP/Basic call/Receipt of	of SAM after invit	e has be	en sent
SIP selection criteria:	NOT PICS 3/2			
ISUP selection criteria:	PICS 3/8			
Test purpose:	Ensure that the SUT in Idle st indicator in the Nature of Concheck not required" on rece	tate, on receipt o nection Indicator ipt of a SAM star	f an IAM s param t Ti/w1. /	of address signalling. Ti/w1 is expired message containing Continuity Check eter which is set to indicate "continuity After Ti/w1 is expired, the SUT: received in the IAM and the SAM.
SIP Parameter	• Serius ari invite mes	sage containing a	all digits	received in the IAW and the SAW.
values:				
ISUP Parameter				
values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM →	Start T <sub>i/w1</sub>		
	SAM →	Start T <sub>i/w1</sub>		
	SAM →	Start T <sub>i/w1</sub>		
	SAM →	Start T <sub>i/w1</sub>		
		T <sub>i/w1</sub> expired	<b>→</b>	INVITE
	ACM(no indication) ←			
	CPG(alerting)		<b>←</b>	180 Ringing
	Ringing tone	е		3 3
	ANM ←		<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conv	ersation	
	<b>→</b>		<b>→</b>	BYE
	RLC +		<del>-</del>	200 OK BYE

TP302005	SIP reference: RFC 3261 [6]			ISUP reference: ES 283 027 [1], clause 7.2.3.2.1a
TSS reference:		ic call/Receipt of SAM after inv	rite has be	een sent
SIP selection criteria:	PICS 3/2			
ISUP selection criteria:	PICS 3/9			
Test purpose:	Overlap proce	dure supported without determ	nining the	end of address signalling
	Check indicate			message containing the Continuity parameter which is set to indicate
	• Send	s an INVITE message start Ti/	w1 and T	i/w2.
	On re	eceipt of a 404/484 the SUT sh	all send a	ACK, stop Ti/w2 and start Ti/w3:
	<ul> <li>On receipt of a SAM from the ISUP the SUT shall send an INVITE request:</li> <li>a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call.</li> <li>b) All other contents of the new INVITE are interworked from the parameters of the original IAM.</li> <li>c) Start Ti/w1 and Ti/w2 and stop Ti/w3 (if it is running).</li> </ul>			
SIP Parameter values:	c) :	<u> </u>	1,,,,,,,	
ISUP Parameter values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
			<b>←</b>	404/484
			<b>→</b>	ACK
	SAM	<b>→</b>	<b>→</b>	INVITE
			<b>←</b>	404/484
			<b>→</b>	ACK
	SAM	<b>→</b>	<b>→</b>	INVITE
			<b>←</b>	404/484
			<b>→</b>	ACK
	SAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b> Ringing tone	<b>←</b>	180 Ringing
	1	+ tinging tene	<del>(</del>	200 OK INVITE
	ANM	•		
	ANM		<b>→</b>	ACK
	ANM			ACK

## 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 mes	ssage		
SIP selection	PICS 3/1			
criteria:				
ISUP selection	PICS 4/9 AND NOT PICS 4/17			
criteria:	1011:	A COLT		
Test purpose:	ACM is sent after the determination of addres	s complete indication in the SUT		
	Ensure that the SUT in Idle state, on receipt of	f an IAM massage containing the complete		
	called party number and the sending compl			
	canca party number and the sending compl	nation.		
	Sends the INVITE message to called use	r and starts Ti/w2.		
	When Ti/w2 is expired, sends the ACM m			
	indication (00)", the Called party's categ			
	"ordinary subscriber (01)" or "payphone (1			
	"interworking encountered (1)", the ISUP	indicator set to "ISUP not used all the way", the		
	ISDN access indicator set to "terminating	g access non-ISDN".		
SIP Parameter				
values:				
ISUP Parameter	IAM; Called party number: complete number			
values:	ACM, CPS indicator: no indication (00)	. (00)		
		ion(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 r	non-ISDN"		
Comments:	ISUP/BICC SUT	SIP		
	IAM →	→ INVITE		
	Ti/w2 expired	2 1111112		
	ACM(no indication)			
	CPG ←	← 180 Ringing		
		ng tone		
	ANM ←	€ 200 OK INVITE		
		→ ACK		
	Conv	ersation		
	<b>→</b>	→ BYE		
	RLC	← 200 OK BYE		
	1120	- LOU GILDIE		

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 mes	ssage
SIP selection	PICS 2/3 AND PICS 3/1	
criteria:		
ISUP selection	PICS 4/9 AND PICS 4/17	
criteria:		
Test purpose:	64 kBit/s unrestricted call, ACM is sent after indication in the SUT	the determination of <b>address complete</b>
	Ensure that the SUT in Idle state, on receipt of called party number and the sending complete.	
	Sends the INVITE message to called use     When Titus is expired, sends the ACM m.	r and starts Ti/w2. essage with the <b>CPS indicator</b> set to "no
	indication (00)", the Called party's categ	ory indicator set to "no indication(00)" or
		10)", the interworking indicator set to "no
		ndicator set to "ISUP used all the way", the
SIP Parameter	ISDN access indicator set to "terminatin INVITE: SDP a=rtpmap: <dynamic-pt> CLEAR</dynamic-pt>	
values:	INVITE: SUP a=ripmap. <uynamic-pt> CLEAR</uynamic-pt>	RIVIODE/8000
ISUP Parameter	IAM; Called party number: complete number	: TMR: "64 kbit/s unrestricted"
values:	ACM, CPS indicator: no indication (00)	,
		ion(00) or ordinary subscriber (01) or payphone
	(10)	
	interworking indicator: no interworking enco	untered (0)
	ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access IS	DN"
Comments:	ISUP/BICC SUT	SIP
Johnnents.	IAM →	→ INVITE
	Ti/w2 expired	
	ACM(no indication)	
	CPG ←	← 180 Ringing
	Ringi	ing tone
	ANM <b>←</b>	← 200 OK INVITE
		→ ACK
		ersation
	<b>→</b>	→ BYE
	RLC <del>C</del>	← 200 OK BYE

TP303003	SIP reference: RFC 3261 [6]	ES 283	ISUP reference: 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	PICS 3/1		
criteria:			
ISUP selection	PICS 4/9 AND NOT PICS 4/17		
criteria:			
Test purpose:	ACM is sent after the maximum number of or received  Ensure that the SUT in Idle state, on receipt on number of digits used in the national number.	f an IAM i	message containing the <b>maximum</b>
	<ul> <li>Sends the INVITE message to the called user and starts Ti/w2.</li> <li>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".</li> </ul>		
SIP Parameter			
values:			
ISUP Parameter values:	IAM; Called party number: complete number ACM, CPS indicator: no indication (00)	•	
values:	Called party's category indicator: no indicator: no indicator:	tion(00) o	r 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	<b>→</b>	SIP INVITE
	Ti/w2 expired	-	INVITE
	ACM(no indication)	I	
	CPG	<b>←</b>	180 Ringing
		ing tone	100 Tanging
	ANM	₩ <b>←</b>	200 OK INVITE
		<b>→</b>	ACK
	Conv	ersation	
	<b>→</b>	<b>→</b>	BYE
	RLC <del>(</del>	<u>+</u>	200 OK BYE

TP303004	SIP reference: RFC 3261 [6]	FS 283 (	ISUP reference: 027 [1], clauses 7.2.3.2.1.4, 7.2.3.2.4
		20 200 (	and 7.2.3.2.5
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message		
SIP selection	PICS 2/3 AND PICS 3/1		
criteria:			
ISUP selection	PICS 4/9 AND PICS 4/17		
criteria:			
Test purpose:	64 kBit/s call, ACM is sent after the maximum numbering plan received		
	Ensure that the SUT in Idle state, on receipt o number of digits used in the national number.		
	<ul> <li>Sends the INVITE message to the called</li> <li>When Ti/w2 is expired, sends the ACM m</li> </ul>		
	indication (00)", the Called party's categ	ory indica	ator set to "no indication(00)" or
	"ordinary subscriber (01)" or "payphone (		
	interworking encountered", the ISUP indi		
SIP Parameter	access indicator set to "terminating accelliNVITE: SDP a=rtpmap: <dynamic-pt> CLEAI</dynamic-pt>		
values:	INVITE: SUP a=riprilap. <uyriamic-pt> CLEAR</uyriamic-pt>	XIVIODE/60	500
ISUP Parameter	IAM; Called party number: complete number		
values:	ACM, <b>CPS indicator:</b> no indication (00)		
	Called party's category indicator: no indicat	ion(00) or	ordinary subscriber (01) or payphone
	(10)		
	interworking indicator: no interworking enco	untered	
	ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access l	SDVI.	
Comments:	ISUP/BICC SUT	אועט	SIP
Joinnetts.	IAM →	<b>→</b>	INVITE
	Ti/w2 expired	2	
	ACM(no indication) ←		
	CPG ←	<b>←</b>	180 Ringing
	Ringi	ng tone	
	ANM <b>←</b>	<b>←</b>	200 OK INVITE
		<b>→</b>	ACK
		ersation	
	<b>→</b>	<b>→</b>	BYE
	RLC ←	<u> </u>	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
<b>700</b> (	10117 017 /7		and 7.2.3.2.5
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message		
SIP selection	PICS 3/1		
criteria:	PICS 4/9 AND NOT PICS 4/17		
ISUP selection criteria:	PICS 4/9 AND NOT PICS 4/17		
Test purpose:	ACM is sent after sufficient number of digits	has hoor	received to route the call to the called
rest purpose.	party.		
	Ensure that the SUT in Idle state, on receipt or called party number where the end of address called party number to indicate that a sufficient route the call to the called party:	ss signallir	ng is determined by analysis of the
	route the can to the canca party.		
	Sends the INVITE message to the called	user and s	starts Ti/w2.
	When Ti/w2 is expired, sends the ACM m		
	indication (00)", the Called party's categ	ory indica	ator set to "no indication(00)" or
	"ordinary subscriber (01)" or "payphone (1		
	"interworking encountered (1)", the <b>ISUP</b>		
OID Davis and the	ISDN access indicator set to "terminatin	g access r	non-ISDN".
SIP Parameter values:			
ISUP Parameter	IAM; Called party number: complete number		
values:	ACM, CPS indicator: no indication (00)		
	Called party's category indicator: no indicat	ion(00) or	ordinary subscriber (01) or payphone
	(10)	(,	(- , - , - , - , - , - , - , - , - , - ,
	interworking indicator: interworking encount	ered (1)	
	ISUP indicator: ISUP not used all the way		
	ISDN access indicator: "terminating access r	non-ISDN"	
Comments:	ISUP/BICC SUT	_	SIP
	IAM →	<b>→</b>	INVITE
	Ti/w2 expired		
	ACM(no indication) ←	_	400 B: :
	CPG ←	. ←	180 Ringing
		ng tone	200 OK INVITE
	ANM ←	<b>←</b>	200 OK INVITE
	Conv	ersation	ACK
	→ Conv	ersation <del>&gt;</del>	BYE
	RLC +	→ ←	200 OK BYE
	INLO T		ZUU UN DIE

TP303006	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	PICS 2/3 AND PICS 3/1		
criteria:			
ISUP selection	PICS 4/9 AND PICS 4/17		
criteria:			
Test purpose:	64 kBit/s unrestricted call, ACM is sent after	sufficier	nt number of digits has been received
	to route the call to the called party		<u>-</u>
	Ensure that the SUT in Idle state, on receipt o		
	called party number where the end of address		
	called party number to indicate that a sufficie	nt numbe	er of digits has been received to
	route the call to the called party:		
	Sends the INVITE message to called use		
	When Ti/w2 is expired, sends the ACM m		
	indication (00)", the Called party's categ		
	"ordinary subscriber (01)" or "payphone (		
	interworking encountered (0)", the ISUP i		
CID Devementes	ISDN access indicator set to "terminatin		
SIP Parameter values:	INVITE: SDP a=rtpmap: <dynamic-pt> CLEAI</dynamic-pt>	KIVIODE/8	3000
ISUP Parameter	IAM; Called party number: complete number	TMD: "G	4 khit/a uprostricted"
values:	ACM, <b>CPS indicator:</b> no indication (00)	, HVIK. O	4 KDII/S UIIIeStricteu
values.	Called party's category indicator: no indicat	ion(00) or	ordinary subscriber (01) or navnhone
	(10)	1011(00) 01	ordinary subscriber (01) or payprione
	interworking indicator: no interworking enco	untered ((	n)
	ISUP indicator: ISUP used all the way	untered (t	5)
	ISDN access indicator: "terminating access	SDN"	
Comments:	ISUP/BICC SUT		SIP
	IAM →	<b>→</b>	INVITE
	Ti/w2 expired	_	
	ACM ←	ı	
	CPG ←	<b>←</b>	180 Ringing
	Ringing tone	•	100 Tanging
	ANM ←	<b>←</b>	200 OK INVITE
	LI AIVI	<b>→</b>	ACK
	Conv	ersation	ACR
	→	ersation	BYE
		_	
	RLC ←	<u>+</u>	200 OK BYE

TP303007	SIP reference: RFC 3261 [6]		ISUP reference:
		ES 283 0	27 [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	PICS 3/1		
criteria:			
ISUP selection	NOT PICS 4/17		
criteria:			
Test purpose:	ACM is sent determined by the expiration time	er T <sub>I/W1</sub>	
	Ensure that the SUT in Idle state, on receipt of called party number where the end of address timer T <sub>I/W1</sub> after the receipt of the latest ad	s signallin	g is determined by the expiration
	<ul> <li>sends the INVITE message to the called to Sends the ACM message with the CPS in party's category indicator set to "no ind "payphone (10)", the interworking indicator ISUP indicator set to "ISUP not used all to "terminating access non-ISDN".</li> </ul>	idicator selection (00)	or "ordinary subscriber (01)" or "interworking encountered (1)", the
SIP Parameter	-		
values:			
ISUP Parameter	IAM; Called party number: complete number		
values:	ACM, CPS indicator: no indication (00)	: (00)	
	Called party's category indicator: no indicat	ion(uu) 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 r	non-ISDN"	
Comments:	ISUP/BICC SUT		SIP
	IAM → Start T <sub>I/W1</sub>		
	T <sub>I/W1</sub> expiry		
	ACM(no indication) ←	<b>→</b>	INVITE
	CPG ←	<b>←</b>	180 Ringing
	Ringing tone		
	ANM <b>←</b>	<b>←</b>	200 OK INVITE
		<b>→</b>	ACK
		ersation	
	<b>→</b>	<b>→</b>	BYE
	RLC ←	+	200 OK BYE

TP303008	SIP reference: RFC 3261 [6]		ISUP reference:
		ES 283 027	7 [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	PICS 2/3 AND PICS 3/1		
criteria:			
ISUP selection	PICS 4/17		
criteria:			
Test purpose:	64 kBit/s unrestricted call, ACM is sent after de	etermined by	the expiration timer T <sub>I/W1</sub>
	Ensure that the SUT in Idle state, on receipt of containing the complete called party number determined by the expiration timer T <sub>I/W1</sub> af	where the e er the recei	nd of address signalling is
	<ul> <li>Sends the INVITE message to called user.</li> <li>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".</li> </ul>		
SIP Parameter	INVITE: SDP a=rtpmap: <dynamic-pt> CLEAF</dynamic-pt>	MODE/8000	)
values:			
ISUP Parameter	IAM; Called party number: complete number	TMR: "64 kl	bit/s unrestricted"
values:	ACM, CPS indicator: no indication (00)	4	
	Called party's category indicator: no indicat	on(00) or or	dinary subscriber (01) or payphone
	(10)		
	interworking indicator: no interworking encountered (0) ISUP indicator: ISUP used all the way		
	ISDN access indicator: "terminating access I	SDN"	
Comments:	ISUP/BICC SUT		IP
	IAM → Start T <sub>IW1</sub>		
	T <sub>I/W1</sub> expiry		
	ACM(no indication) ←	<b>→</b> IN	IVITE
	CPG ←		80 Ringing
	Ringing tone		5 ·gg
	ANM ←	<b>←</b> 20	00 OK INVITE
		<b>→</b> A	CK
	Conv	ersation	
	<b>→</b>	<b>→</b> B	YE
	RLC <b>←</b>	← 2	00 OK BYE

TP303009	SIP reference: RF	C 3261 [6]	ES 2	ISUP reference: 283 027 [1], clauses 7.2.3.2.5 and 7.2.3.2.1.4
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection	PICS 3/2			
criteria:				
ISUP selection	NOT PICS 4/17			
criteria:				
Test purpose:	ACM is sent determined by	y the expiration time	<sup>r T</sup> I/W2	
	an IAM message containin has been received (start tir procedure):  Sends an INVITE mess party's category indi "payphone (10)", the in	g the minimum numer TI/W2 and involute sage to the called upage with the CPS in cator set to "no indinterworking indica	ser and a dicator s cation(00 tor set to	toward the SIP network, on receipt of digits required for routing the call propriate outgoing SIP signalling after the expiration of T <sub>I/W2</sub> .  Set to "no indication (00)", the Called of "or "ordinary subscriber (01)" or "interworking encountered (1)", the
			ne way",	the ISDN access indicator set to
SIP Parameter values:	"terminating access no	JII-ISDIN .		
ISUP Parameter	IAM; Called party number: complete number			
values:	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 →	- 10.1.1 1/VV I		
	SAM →	Ctart 1////		
	SAM →	Start T <sub>I/W2</sub> T <sub>I/W2</sub> expiry	<b>→</b>	INVITE
	ACM(no indication) ←			
	CPG		<b>←</b>	180 Ringing
		•	ng tone	
	ANM ←		<del>(</del>	200 OK INVITE
		_	→	ACK
	_		ersation	
	<b>→</b>		<b>→</b>	BYE
	RLC +		+	200 OK BYE

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	PICS 2/3 AND PICS 3/2			
criteria:				
ISUP selection	PICS 4/17			
criteria:				
Test purpose:	64 kBit/s unrestricted call, ACM is sent after de	determined by the expiration timer T <sub>I/W2</sub>		
	Ensure that the SUT if overlap addressing is to an IAM message, TMR=64 kBit/s unrestricted required for routing the call has been recei appropriate outgoing SIP signalling procedure  Sends an INVITE message to the called up	ived (start timer TOIW2 and invoke the e):		
	party's category indicator set to "no ind "payphone (10)", the interworking indicator ISUP indicator set to "ISUP used all the "terminating access ISDN".			
SIP Parameter	INVITE: SDP a=rtpmap: <dynamic-pt> CLEAF</dynamic-pt>	RMODE/8000		
values:				
ISUP Parameter	IAM; Called party number: complete number, TMR: "64 kbit/s unrestricted"			
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: "terminating access I	ISDN"		
Comments:	ISUP/BICC SUT	SIP		
	IAM → Start T <sub>I/W1</sub>			
	SAM → Start T <sub>I/W1</sub>			
	SAM → Start T <sub>I/W2</sub>	→ INVITE		
	T <sub>/IW2</sub> expiry	,		
	ACM(no indication) ←			
	CPG ←	← 180 Ringing		
	Ringing tone			
	ANM <b>←</b>	← 200 OK INVITE		
		→ ACK		
		versation		
	<b>→</b>	→ BYE		
	RLC ←	← 200 OK BYE		

TP303011	SIP reference: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection	PICS 3/1			
criteria:				
ISUP selection	NOT PICS 4/9 AND NOT PICS 4/17			
criteria:				
Test purpose:	ACM is sent after 180 Ringing was received			
	Ensure that the SUT in Idle state, on receipt of called party number, on receipt of a 180 Ring	ging mes	sage:	
	<ul> <li>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".</li> </ul>			
SIP Parameter				
values:				
ISUP Parameter	IAM; Called party number: complete number			
values:	ACM, CPS indicator: subscriber free (01)	(0.0)	" (04)	
	Called party's category indicator: no indicati	on(00) o	r ordinary subscriber (01) or payphone	
	(10) interworking indicator: interworking encounter	arad (1)		
	ISUP indicator: ISUP not used all the way	=1 <b>e</b> u (1)		
	ISDN access indicator: "terminating access r	on-ISDN	u	
Comments:	ISUP/BICC SUT		SIP	
	IAM →	<b>→</b>	INVITE	
	ACM ←	<b>←</b>	180 Ringing	
	Ringing tone		3 3	
	ANM ←	<b>←</b>	200 OK INVITE	
		<b>→</b>	ACK	
	Conversation			
	<b>→</b>	<b>→</b>	BYE	
	RLC ←	<b>←</b>	200 OK BYE	

TP303012	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.2.	5	
TSS reference:	_ <del></del>			
SIP selection	ISUP-SIP /Basic call/Sending of the ACM mes	age		
criteria:	FICS 2/3 AIND FICS 3/1			
ISUP selection	NOT PICS 4/9 AND PICS 4/17			
criteria:	1100 4/37/1100 4/17			
Test purpose:	64 kBit/s unrestricted call, ACM is sent after 1	0 Ringing was received		
	Ensure that the SUT in Idle state, on receipt of containing the complete called party number	on receipt of a 180 Ringing message:		
	<ul> <li>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".</li> </ul>			
SIP Parameter	INVITE: SDP a=rtpmap: <dynamic-pt> CLEAF</dynamic-pt>	MODE/8000		
values:				
ISUP Parameter	IAM; Called party number: complete number	TMR: "64 kbit/s unrestricted"		
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: "terminating access ISDN"			
Comments:	ISUP/BICC SUT	SIP		
	IAM →	→ INVITE		
	ACM ←	← 180 Ringing		
	Ringing tone			
	ANM ←	← 200 OK INVITE		
	_	→ ACK		
	Conversation			
	<b>→</b>	→ BYE		
	RLC ←	<b>←</b> 200 OK BYE		

TP303013	SIP reference: RFC 3261 [6]	E	ISUP reference: IS 283 027 [1], clause 7.2.3.2.4	
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message			
SIP selection	PICS 4/15			
criteria:				
ISUP selection	NOT PICS 4/9 AND NOT PICS 4/17			
criteria:				
Test purpose:	The SUT supports the P-Early-Media header			
SIP Parameter values:	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 "subscriber free (01)", Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", OBCI "in -band information" set to: yes.			
ISUP Parameter	IAM; Called party number: complete numbe	r		
values:	ACM; CPS indicator: subscriber free (01),			
	OBCI: in -band information: yes			
Comments:	ISUP/BICC SUT		SIP	
	IAM →	<b>→</b>	INVITE	
	ACM	<b>←</b>	180 Ringing	
	ANM ←	+	200 OK INVITE	
		<b>→</b>	ACK	
	Conv	ersation/		
	<b>→</b>	<b>→</b>	BYE	
	RLC +	+	200 OK BYE	

TP303014	SIP refere	ence: RFC 3261 [6]	F	ISUP reference: S 283 027 [1], clause 7.2.3.2.5		
TSS reference:	ISUP-SIP /Basic o	call/ Sending of the ACM me				
SIP selection	PICS 4/18	bally containing of the Activitino	bougo			
criteria:						
ISUP selection						
criteria:						
Test purpose:	180 received, ma	pping of PSTN XML Progres	sIndicato	r #7 into the ACM BCI		
	<ul> <li>Ensure that the SUT, if an ACM has not been already sent, on receipt the 180 Ringing message, with the PSTN XML body with ProgressIndicator # 7 (Terminating user ISDN):</li> <li>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".</li> </ul>					
SIP Parameter	180 Ringing;					
values:	PSTN XML body	with ProgressIndicator # 7 (T	erminatir	ng user ISDN)		
ISUP Parameter	ACM. CPS indica	itor: subscriber free (01)				
values:	Called party's ca		ion(00) oı	ordinary subscriber (01) or payphone		
	(10) interworking indicator: no interworking encountered (0) ISUP indicator: ISUP used all the way ISDN access indicator: terminating access is ISDN access delivery information: Set-up message generated (IF PRESENT)					
Comments:	ISUP/BICC	SUT		SIP		
	IAM	<b>→</b>	_	15 N (775		
	4014		<b>→</b>	INVITE		
	ACM	Einging tone	<b>←</b>	180 Ringing		
	ANM	Ringing tone	<b>←</b>	200 OK INVITE		
	LZI AIAI	•	→	ACK		
		Conv	ersation	71011		
		<b>→</b>	- Sation →	BYE		
	RLC	<b>+</b>	<b>←</b>	200 OK BYE		

TP303015	SIP refe	rence: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic	call/ Sending of the ACM me	ssage		
SIP selection criteria:	PICS 4/18				
ISUP selection					
criteria:					
Test purpose:	180 received, ma the ATP in the so		ssIndicato	r into Progress Indicator contained in	
	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				
	ProgressIndicato	or #7:			
				or set to " subscriber free (01)" and the g the progress indicator PI_VALUE.	
SIP Parameter	180 Ringing;				
values:		gressIndicator: PI_VALUE (I	PIXIT)		
ISUP Parameter		ator: subscriber free (01)			
values:		dicator: PI_VALUE (PIXIT)			
Comments:	ISUP/BICC	SUT		SIP	
	IAM	<b>→</b>			
			<b>→</b>	INVITE	
	ACM	<b>←</b>	<b>←</b>	180 Ringing	
		Ringing tone			
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE	
			<b>→</b>	ACK	
		Conv	ersation		
		<b>→</b>	<b>→</b>	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]	E	ISUP reference: S 283 027 [1], clause 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic call/Se	ending of the ACM mes	sage		
SIP selection criteria:	PICS 3/1 AND NOT PI	CS 4/15			
ISUP selection criteria:	NOT PICS 4/9 AND NO	OT PICS 4/17			
Test purpose:  SIP Parameter values:	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.				
ISUP Parameter values:	IAM; Called party num	iber. complete number			
Comments:	ISUP/BICC	SUT		SIP	
	IAM	<b>→</b>	<b>→</b>	INVITE 183 Session Progress	
	ACM	<del>(</del>	<b>←</b>	180 Ringing	
		Ringi	ng tone		
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE	
			<b>→</b>	ACK	
			ersation		
		<b>→</b>	<b>→</b>	BYE	
	RLC	+	+	200 OK BYE	

TP303017	SIP reference: RFC 3261 [6]		E	ISUP reference: S 283 027 [1], clause 7.2.3.2.5	
TSS reference:	ISUP-SIP /Basic call/Sending of the A	CM messa	ge		
SIP selection	PICS 4/15				
criteria:					
ISUP selection					
criteria:					
Test purpose:	P-Early-Media header supported, 183	3 is interwor	ked, aı	n 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 <b>CPS indicator</b> set to " no indication (00)", the <b>Called party's category indicator</b> set to "no indication(00)", OBCI "in -band information" set to: ves.				
SIP Parameter	183 Session Progress that includes a	P-Early-Me	edia he	ader authorizing early media	
values:					
ISUP Parameter	IAM; Called party number: complete				
values:	ACM; CPS indicator: no indication (0	00),			
Comments:	OBCI: in -band information: yes	SUT		SIP	
Comments:	IAM →	501	<b>→</b>	INVITE	
	ACM(no indication)		<del>-</del>	183 Session Progress	
	CPG(Alerting)		<del>-</del>	180 Ringing	
	Ringing tone		_	160 Kinging	
	ANM ←		<b>←</b>	200 OK INVITE	
	\(\text{VIAINI}\)		<b>→</b>	ACK	
		Convers	_	AUN	
	<b>→</b>	COLIVEIS	•alion	BYF	
	RLC ←		<del>-</del>	200 OK BYE	
	INLO			200 ON DTL	

TP303018	SIP referen	ice: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 9.2.3.3.12		
TSS reference:	ISUP-SIP /Basic ca	III/ Sending of the ACM me	ssage			
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	<b>→</b>	<b>→</b>	INVITE		
			<b>←</b>	183 Session Progress		
	ACM	← Ringing tone	<b>←</b>	180 Ringing		
	ANM	<b>~</b>	<b>←</b>	200 OK INVITE		
			<b>→</b>	ACK		
		Conv	ersation			
		<b>→</b>	<b>→</b>	BYE		
	RLC	<b>←</b>	+	200 OK BYE		

TP303019	SIP refer	ence: RFC 3261 [	6]	E	ISUP reference: S 283 027 [1], clause 9.2.3.3.12	
TSS reference:	ISUP-SIP /Basic	call/ Sending of th	e ACM mes	ssage		
SIP selection criteria:	PICS 3/1			<u> </u>		
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	<b>→</b>		<b>→</b>	INVITE	
				<b>←</b>	183 Session Progress	
	ACM	<b>←</b>		<del>(</del>	180 Ringing	
		Ringing tone			3 3	
	ANM	<b>←</b>		+	200 OK INVITE	
	1			<b>→</b>	ACK	
			Conve	ersation		
		<b>→</b>	001100	→	BYE	
	RLC	<b>←</b>		<b>←</b>	200 OK BYE	

TP303020	SIP reference: RF	C 3261 [6]	E	ISUP reference: 5 283 027 [1], clause 9.2.3.3.12		
				5 203 027 [1], Clause 9.2.3.3.12		
TSS reference:	ISUP-SIP /Basic call/ Send	ling of the ACM mes	ssage			
SIP selection	PICS 3/1 NOT PICS 4/15					
criteria:						
ISUP selection	NOT PICS 4/9					
criteria:						
Test purpose:	183 received after T <sub>I/W1</sub> ex	kpired, P-Early-Med	ia headei	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 <sub>I/W1</sub> after the receipt of the latest address message on receipt of a 183 Session Progress:					
SIP Parameter	No ISUP message is s	eni backwaru.				
values:						
ISUP Parameter						
values:						
Comments:	ISUP/BICC	SUT		SIP		
	IAM →					
		T <sub>I/W1</sub> expiry				
	ACM(no indication) ←		<b>→</b>	INVITE		
	,		<b>←</b>	183 Session Progress		
	CPG(alerting) ←		<b>←</b>	180 Ringing		
	, ,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,	ng tone		3 3		
	ANM		<b>←</b>	200 OK INVITE		
			<b>→</b>	ACK		
		Conve	ersation	-		
	→	30	<b>→</b>	BYE		
	RLC ←		<del>-</del>	200 OK BYE		

TP303021	SIP reference: RFC 3261 [6]	ES	ISUP reference: 283 027 [1], clause 7.2.3.2.5			
T00 (	10115 015 /5 : 11/0 15 (11 4 014	1	203 027 [1], Clause 7.2.3.2.3			
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM me	ssage				
SIP selection criteria:	PICS 4/15 AND PICS 4/18					
ISUP selection criteria:						
Test purpose:	183 received, mapping of PSTN XML Progres	alndiaatar t	HT into the ACM BCI			
rest purpose.	103 received, mapping of F3 IN XIVIL Frogres	ssiriuicator <del>r</del>	+7 Into the ACM BCI			
	Ensure that the SUT, on receipt of the 183 Se ProgressIndicator # 7 (Terminating user ISE	N):	-			
	<ul> <li>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".</li> </ul>					
SIP Parameter values:	183 Session Progress; PSTN XML Progress	Indicator #	7 (Terminating user ISDN)			
ISUP Parameter	ACM, CPS indicator: no indication (00)					
values:	Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone					
	(10)	` ,	, , , ,			
	interworking indicator: no interworking enco	ountered (0)				
	ISUP indicator: ISUP used all the way					
	ISDN access indicator: ISDN					
	access delivery information: Set-up messa					
Comments:	ISUP/BICC SUT		SIP			
	IAM →	=	INVITE			
	ACM ←		183 Session Progress			
	CPG ←	+	180 Ringing			
	Ringing tone	_				
	ANM ←		200 OK INVITE			
			ACK			
		ersation				
	<b>→</b>		BYE			
	RLC ←	<del></del>	200 OK BYE			

TP303022	SIP refer	ence: RFC 3261 [6]		ISUP reference: S 283 027 [1], clause 7.2.3.2.5		
TCC materials	ISUP-SIP /Basic call/Sending of the ACM message					
TSS reference: SIP selection	PICS 4/15 AND F	call/Sending of the ACM me	ssage			
criteria:	PICS 4/15 AND I	PICS 4/18				
ISUP selection						
criteria:						
Test purpose:	the ATP in the se	ent ACM		r into Progress Indicator contained in		
	body with Progre	ss indicator # 7 (Terminating	user ISD	gress message with the PSTN XML N) containing the <b>PSTN XML</b> ssIndicator #7 is not interworked:		
	<ul> <li>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".</li> </ul>					
SIP Parameter values:	183 Session Prog	gress; <b>PSTN XML Progress</b>	Indicator	: PI_VALUE		
ISUP Parameter	ACM: CPS indic	ator: no indication (00)				
values:		ategory indicator: no indica	tion(00) o	r ordinary subscriber (01) or payphone		
	(10)			_,		
	interworking indicator: no interworking encountered (0)					
		ISUP used all the way				
	ISDN access inc	ndicator: ISDIN				
		information: Set-up messa	ne nenera	ted (IF PRESENT)		
Comments:	ISUP/BICC	SUT	go gonora	SIP		
3	IAM	<b>→</b>	<b>→</b>	INVITE		
	ACM	<b>←</b>	<b>←</b>			
	CPG	<del>-</del>	<del>(</del>	180 Ringing		
		Ringing tone		3 3		
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE		
			<b>→</b>	ACK		
		Conv	ersation			
		<b>→</b>	<b>→</b>	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 r	eference: RFC 3261 [6]	F	ISUP reference: S 283 027 [1], clause 7.2.3.2.5			
TSS reference:	ISUP-SIP /Basic	ISUP-SIP /Basic call/Sending of the ACM message					
SIP selection		AND NOT PICS 4/18	sage				
criteria:	11011100 4/10	7.11D 1100 4/10					
ISUP selection							
criteria:							
Test purpose:	183 received, mapping of PSTN XML ProgressIndicator into Progress Indicator contained in the ATP and mapping of PSTN XML ProgressIndicator #7 BCI is 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	183 Session Pro	ogress; PSTN XML Progressi	ndicator:	PI VALUE			
values:				_			
ISUP Parameter							
values:							
Comments:	ISUP/BICC	SUT		SIP			
	IAM	<b>→</b>	<b>→</b>	INVITE			
			<b>←</b>	183 Session Progress			
	ACM	<b>←</b>	<b>←</b>	180 Ringing			
		Ringing tone					
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE			
			<b>→</b>	ACK			
		Conv	ersation				
		<b>→</b>	<b>→</b>	BYE			
	RLC	<b>←</b>	<b>←</b>	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				
T00 (	10110 010 /0 : 11/0 1: (11 4 014	= =:				
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM me	ssage				
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 4/1	7				
Test purpose:	Preconditions requested, ACM is sent after the in the SUT  Ensure that the SUT in Idle state, on receipt of	ne determination of address complete indication				
	called party number, the sending complete on this circuit (ISUP) or COT is expected (BIO	e indication, and the continuity check is required CC):				
	received.	til a successful continuity indication has been				
	<ul> <li>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".</li> </ul>					
SIP Parameter values:						
ISUP Parameter	IAM; Called party number: complete number	r				
values:	ACM, <b>CPS indicator:</b> no indication (00)	·				
	Called party's category indicator: no indication (00) or ordinary subscriber (01) or payphone					
	(10)					
	interworking indicator: interworking encoun	tered (1)				
	ISUP indicator: ISUP used all the way					
	ISDN access indicator: "terminating access	non-ISDN"				
Comments:	ISUP/BICC SUT	SIP				
	IAM →	→ INVITE				
		<ul> <li>183 Session Progress</li> </ul>				
		→ PRACK				
		← 200 OK PRACK				
	COT →	→ UPDATE				
		← 200 OK UPDATE				
	ACM ← Ti/w2 expired	d l				
	CPG(Alerting) ←	← 180 Ringing				
	( - · · · · · · · · · · · · · · · · · ·	→ PRACK				
		← 200 OK PRACK				
	Ringing tone					
	ANM •	← 200 OK INVITE				
	/ · · · · · · · · · · · · · · · · · · ·	→ ACK				
	Con	versation				
	→	→ BYE				
	RLC <del>C</del>	€ 200 OK BYE				
	INLO <b>T</b>	<b>1</b> 200 ON D1E				

TP303025	SIP reference:	: RFC 3261 [6]		ISUP reference:	
			Е	S 283 027 [1], clause 7.3.3.2.3	
TSS reference:		ending of the ACM mes	sage		
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, Precondition indication in the SUT	s requested, ACM is se	nt after th	ne determination of address complete	
	called party number,		indication	message containing the complete n and the continuity check is required	
	<ul> <li>The SUT shall with received.</li> </ul>	· ·	l a succe	ssful continuity indication has been	
	<ul> <li>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".</li> </ul>				
SIP Parameter values:	Ŭ	p: <dynamic-pt> CLEAR</dynamic-pt>	RMODE/8	8000	
ISUP Parameter	IAM: Called party nur	nber: complete number	. TMR: "6	4 kbit/s unrestricted"	
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)				
		r: no interworking enco	untered (	0)	
	ISUP indicator: ISUP		CDNI"		
Comments:	ISUP/BICC	or: "terminating access I SUT	2DIN	SIP	
Comments:	IAM	301 →	<b>→</b>	INVITE	
	IAW	7	→ ←		
			<b>→</b>	183 Session Progress PRACK	
			=		
	007	•	<del>(</del>	200 OK PRACK	
	СОТ	<b>→</b>	<b>→</b>	UPDATE	
			+	200 OK UPDATE	
	ACM(no indication)	← Ti/w2 expired	_		
	CPG	<b>←</b>	+	3 3	
			<b>→</b>	PRACK	
			+	200 OK PRACK	
	ANM	<b>←</b>	<b>←</b> →	200 OK INVITE ACK	
		Conv	=	ACK	
			ersation	BYE	
	DI C	<del>)</del>	<b>→</b>		
	RLC	+	<u> </u>	200 OK BYE	

TP303026	SIP reference	ce: RFC 3261 [6]		ISUP reference:		
				S 283 027 [1], clause 7.3.3.2.3		
TSS reference:		I/Sending of the INVITE r	nessage			
SIP selection criteria:		3 4/5 AND PICS 4/11				
ISUP selection criteria:	PICS 4/2 AND NOT	PICS 4/17				
Test purpose:	Preconditions reque	ested, ACM is sent after e	xpiration o	f timer T <sub>I/W2</sub>		
	Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the <b>minimum number of digits required for routing the call</b> 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 vertice received.	withhold sending ACM un	til a succe	ssful continuity indication has been		
	<ul> <li>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".</li> </ul>					
SIP Parameter values:		nap: <dynamic-pt> CLEA</dynamic-pt>	RMODE/8	0000		
ISUP Parameter	IAM; Called party n	umber: complete numbe	r			
values:	ACM, CPS indicator: no indication (00)					
	Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone					
	(10)					
	interworking indica	ator: interworking encoun	tered (1)			
		JP not used all the way				
	ISDN access indica	ator: "terminating access	non ISDN	II .		
Comments:	ISUP/BICC	SUT		SIP		
	IAM	<b>→</b>	<b>→</b>	INVITE		
			<b>←</b>	183 Session Progress		
			<b>→</b>	PRACK		
			<b>←</b>	200 OK PRACK		
	COT	<b>→</b>	<b>→</b>	UPDATE		
			<b>←</b>	200 OK UPDATE		
		T <sub>I/W2</sub> expiry				
	ACM(no indication)					
	CPG(Alerting)	<del>-</del>	<b>←</b>	180 Ringing		
	or o(ruorung)	_	<b>→</b>	PRACK		
			<b>É</b>	200 OK PRACK		
		Ringing tone	•	200 01(110(0))		
	ANM	tringing tone	<b>←</b>	200 OK INVITE		
	/ M VIVI	•	<b>→</b>	ACK		
		Can	ersation	AON		
		→	rersation	BYE		
	DI C	<del>7</del> ←				
	RLC			200 OK BYE		

TP303027	SIP ref	ference: RFC 3261 [6]		ISUP reference:			
				S 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	1 is sent aft	er expiration of timer T <sub>I/W2</sub>			
	Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the <b>minimum number of digits required for routing the call</b> 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 received.	shall withhold sending ACM u	ntil a succe	ssful continuity indication has been			
	<ul> <li>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".</li> </ul>						
SIP Parameter values:		a=rtpmap: <dynamic-pt> CLE</dynamic-pt>	ARMODE/8	3000			
ISUP Parameter	IAM; Called pa	arty number: complete numb	er, TMR: "6	64 kbit/s unrestricted"			
values:		dicator: no indication (00)	,				
			ation(00) o	r ordinary subscriber (01) or payphone			
	(10)						
		indicator: no interworking en	countered (	0)			
		r: ISUP used all the way					
		indicator: "terminating acces	s ISDN"				
Comments:	ISUP/BICC	SUT	_	SIP			
	IAM	<b>→</b>	<b>→</b>	INVITE			
			<u> </u>	183 Session Progress			
			<b>→</b>	PRACK			
		_	<b>←</b>	200 OK PRACK			
	СОТ	→	<b>→</b>	UPDATE			
			+	200 OK UPDATE			
	T <sub>I/W2</sub> expiry						
	ACM	<b>←</b>					
	CPG	<b>←</b>	<b>←</b>	180 Ringing			
			→	PRACK			
			<b>←</b>	200 OK PRACK			
		Ringing tone	-				
	ANM	←	<b>←</b>	200 OK INVITE			
		-	÷	ACK			
		Col	nversation	7.0.1			
		<b>→</b>	• • • • • • • • • • • • • • • • • • •	BYE			
	RLC	<del>-</del>	<del>-</del>				
	INLU	<b>T</b>		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 ACM me:	ssage			
SIP selection criteria:	PICS 4/5 AND PICS 4/11				
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9 AND NOT PICS	5 4/17			
Test purpose:	180 received after preconditions met, an ACM	1 is sent			
	Ensure that the SUT in Idle state, on receipt called party number, the continuity check is of a 180 Ringing message:	of an IAM message containing the complete required on this circuit (ISUP) indication receipt			
	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	IAM; Called party number: complete numbe	r			
values:	ACM, CPS indicator: "subscriber free (01)"				
	Called party's category indicator: no indica	tion(00) or ordinary subscriber (01) or payphone			
	(10)				
	interworking indicator: interworking encoun	tered (1)			
	ISUP indicator: ISUP used all the way				
	ISDN access indicator: "terminating access				
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			
	Conv	versation			
	<b>→</b>	→ BYE			
	RLC +	€ 200 OK BYE			
	INLO	200 OILDIL			

TP303029	SIP reference: RFC 3261 [6]		ISUP reference:		
		E	S 283 027 [1], clause 7.3.3.2.3		
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM mes	sage			
SIP selection criteria:	PICS 2/3 AND PICS 4/5 AND PICS 4/11				
ISUP selection	PICS 4/2 AND NOT PICS 4/9 AND PICS 4/17				
criteria:					
Test purpose:	64 kBit/s call, 180 received after preconditions	met, an	ACM is sent		
	Ensure that the SUT in Idle state, on receipt or called party number, the continuity check is respected (BICC) indication on receipt of a 180	required c	on this circuit (ISUP) or COT is		
	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	INVITE: SDP a=rtpmap: <dynamic-pt> CLEAR</dynamic-pt>				
values:					
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 →	<b>→</b>	INVITE		
		<b>←</b>	183 Session Progress		
		<b>→</b>	PRACK		
		<b>←</b>	200 OK PRACK		
	COT →	<b>→</b>	UPDATE		
		+	200 OK UPDATE		
	ACM <b>←</b>	<del>-</del>	180 Ringing		
		<b>→</b>	PRACK		
		<b>+</b>	200 OK PRACK		
	Ringing tone	_	200 011 10 1011		
	ANM •	<b>←</b>	200 OK INVITE		
		À	ACK		
	Conv	ersation	,,,,,		
	→	••••••••••••••••••••••••••••••••••••••	BYE		
	RLC <b>←</b>	<del>-</del>	200 OK BYE		

TP303030	SIP reference	ce: RFC 3261 [6]	F	ISUP reference: S 283 027 [1], clause 7.2.3.2.5		
TSS reference:	ISLID-SID /Basic cal	I/Sending of the ACM mes		0 200 021 [1]; 010000 112101210		
SIP selection		4/18 AND PICS 4/19	saye			
criteria:	1 100 2/0 / ((100	7-7-10 7 (ND 1-100-47-10				
ISUP selection						
criteria:						
Test purpose:	parameter sent in th	e ACM		ined in the 180 into the TMU		
	Ensure on receipt of a 180 Ringing contains PSTN XML ProgressIndicator #7 and PSTN XML BearerCapability BC_VALUE, an ACM is sent containing the <b>TMU Parameter BC_VALUE</b> . The BCI is set to: ISUP indicator: ISUP is used all the way ISDN access indicator: ISDN					
		or: Interworking not encou	ntered			
SIP Parameter	INVITE;	<u> </u>				
values:		arer Capability: INVITE _				
		Bearer Capability: INVIT	_	7		
ISUP Parameter values:	IAM; USI: Speech/audio3Kbit/s, G.711 A-law [37] USI prime: Unrestr. Digital info T/A, G.711 A-law [36] TMR: 64 kbit/s preferred TMR prime: Speech/audio3Kbit/s					
	ACM: ISLIP indicate	r: ISUP is used all the way	,			
	ISDN access indicate		7			
	Interworking indicator: Interworking not enountered TMU: TMU_VALUE					
Comments:	ISUP/BICC	SUT		SIP		
	IAM	<b>→</b>	<b>→</b>	INVITE		
	ACM	<b>←</b>	<b>←</b>	180 Ringing		
		Ringing tone	_			
	ANM	<b>←</b>	<del>(</del>	200 OK INVITE		
		^	<b>→</b>	ACK		
			ersation	DVE		
	DI C	<b>→</b>	<b>→</b>	BYE		
	RLC	~	<b>+</b>	200 OK BYE		

TP303031	SIP reference: RFC 3261 [6]		ISUP reference:			
		E	S 283 027 [1], clause 7.2.3.2.5			
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM	message				
SIP selection	PICS 2/3 AND PICS 4/18 AND PICS 4/19					
criteria:						
ISUP selection						
criteria:	Manning of DCTN VM DagrayCanability	Jamant sant	ained in the 102 into the TMI			
Test purpose:	Mapping of PSTN XML PearerCapability e parameter sent in the ACM	iemeni conta	amea in the 163 into the 11vio			
	parameter sent in the AGW					
	Ensure on receipt of a 183 Session Progre	ss contains	PSTN XML ProgressIndicator #7 and			
	PSTN XML BearerCapability BC_VALUE,					
	BC_VALUE.		-			
	The BCI is set to:					
	ISUP indicator: ISUP is used all the way					
	ISDN access indicator: ISDN	austarad				
SIP Parameter	Interworking indicator: Interworking not en INVITE;	Junierea				
values:	PSTN XML first Bearer Capability: INVII	F BC1				
values.	PSTN XML second Bearer Capability: IN					
	Total Amil Cooling Louisi Capability:	502				
	183 Session Progress PSTN XML BC: BC	_VALUE and	I XML PI #7			
ISUP Parameter	IAM; USI: Speech/audio3Kbit/s, G.711 A-la					
values:	<b>USI</b> 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	way				
	Interworking indicator: Interworking not en	ountered				
	TMU: TMU_VALUE					
Comments:	ISUP/BICC SUT		SIP			
	IAM →	<b>→</b>				
	ACM(no indication) ←	<b>←</b>				
	ACM ←	+	180 Ringing			
	Ringing tone	_				
	ANM ←	<del>(</del>	200 OK INVITE			
		<b>→</b>	ACK			
		onversation	DVE			
	<b>→</b>	<b>→</b>	BYE			
	RLC +	+	200 OK BYE			

	Values and selection criteria for test purpose TP303030 and TP303031			
Test	ACM Parameter values	18x Provisional	INVITE parameter value	
purposes		response values:		
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	e: RFC 3261 [6]		ISUP reference:	
			E	S 283 027 [1], clause 7.2.3.2.6	
TSS reference:	ISUP-SIP /Basic call/	Sending of the CPG mes	sage		
SIP selection	PICS 3/1				
criteria:					
ISUP selection	PICS 4/9				
criteria:					
Test purpose:	180 received, a CPG	is sent when an ACM wa	s sent be	efore	
	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 <b>event indicator</b> set to "Alerting".				
SIP Parameter				3	
values:					
ISUP Parameter					
values:					
Comments:	ISUP/BICC	SUT		SIP	
	IAM	<b>→</b>	<b>→</b>	INVITE	
		Ti/w2 expired			
	ACM(no indication)	<b>←</b>			
	CPG(Alerting)	<b>←</b>	<b>←</b>	180 Ringing	
	Ringing tone				
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE	
			<b>→</b>	ACK	
		Conve	ersation		
		<b>→</b>	<b>→</b>	BYE	
	RLC	<b>←</b>	<b>←</b>	200 OK BYE	

TP304002	SIP reference	: RFC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.2.6	
TSS reference:	ISUP-SIP /Basic call/ \$	Sending of the CPG mes	ssage		
SIP selection criteria:	PICS 3/1				
ISUP selection criteria:	NOT PICS 4/15				
Test purpose:	ACM was sent after T <sub>I/W1</sub> expiry, a 183 is not interworked  Ensure that the SUT, having sent a ACM message with called party status "no indication" after T <sub>I/W1</sub> expiry, on receipt of a 183 Session progress message:  • ISUP message is sent backward.				
SIP Parameter	1001 moodago io	oon baokwara.			
values:					
ISUP Parameter values:					
Comments:	ISUP/BICC	SUT		SIP	
	IAM	<b>→</b>			
		T <sub>I/W1</sub> expiry			
	ACM	<b>←</b>	<b>→</b>	INVITE	
			<del>(</del>	183 Session Progress	
	Ringing tone				
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE	
			<b>→</b>	ACK	
			ersation		
		<b>→</b>	<b>→</b>	BYE	
	RLC	+	<u>←</u>	200 OK BYE	

TP304003	SIP reference	RFC 3261 [6]	_	ISUP reference:			
TSS reference:	ES 283 027 [1], clause 7.2.3.1.4  ISUP-SIP /Basic call/ Sending of the CPG message						
SIP selection	PICS 3/1 AND PICS 4		ssage				
criteria:	PICS 3/1 AND PICS 4	/18 AND PICS 4/19					
ISUP selection							
criteria:							
Test purpose:	ACM was sent after $T_{l}$	$_{ m W1}$ expiry, a CPG is se	nt when a	183 is received contains a PSTN XML			
	ProgressIndicator #7:						
				er the expiry of ToiW1, on receipt of ressIndicator "Terminating access			
	sends the CPG message with the <b>event indicator</b> set to "progress".						
SIP Parameter	183 Session Progress						
values:		ndicator "Terminating a	access IS	DN"(#7)			
ISUP Parameter	CPG; event indicator	: progress					
values:	BCI			_,			
	interworking indicator: no interworking encountered (0)						
	ISUP indicator: ISUP	,	D. I.I.				
Commonto		: "terminating access IS	DN"	CID			
Comments:	ISUP/BICC	SUT		SIP			
	IAM	<b>→</b>					
		T <sub>I/W1</sub> expiry					
	ACM	<b>←</b>	<b>→</b>				
	CPG(Progress)	<del>-</del>	<b>←</b>	183 Session progress			
	CPG(Alerting)	<del>-</del>	<b>←</b>	180 Ringing			
	Ringing tone						
	ANM						
			→	ACK			
		Conve	ersation				
		<b>→</b>	<b>→</b>	BYE			
	RLC	<b>←</b>	<b>←</b>	200 OK BYE			

TP304004	SIP reference: RFC 3261 [6]		ISUP reference:		
		E	S 283 027 [1], clause 7.2.3.1.4		
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message				
SIP selection	PICS 3/1 AND PICS 4/18				
criteria:					
ISUP selection					
criteria:					
Test purpose:	ACM was sent after, a 183 contains a PSTN	XML Prog	ressIndicator #7 was received		
	Ensure that the SUT, having sent the ACM message after the reception of the 183 Session progress message, with <b>PSTN XML ProgressIndicator</b> " <i>Terminating access ISDN</i> "(#7), on				
	receipt of an 180 Ringing message, with PS				
	descriptions "Terminating access ISDN"(#7)		dy containing the progress		
	Terrimating access 1901 (#1).				
	• sends the CPG message with the <b>event indicator</b> set to "Alerting".				
SIP Parameter	183 Session Progress: ProgressIndicator "	Terminatin	g access ISDN"(#7)		
values:	180 Ringing: ProgressIndicator "Termination	ng access I	SDN"(#7)		
ISUP Parameter	CPG; event indicator: Alerting				
values:	BCI				
		interworking indicator: no interworking encountered (0)			
	ISUP indicator: ISUP used all the way				
	ISDN access indicator: "terminating access	SDN"			
Comments:	ISUP/BICC SUT		SIP		
	IAM →	<b>→</b>	INVITE		
	ACM <b>←</b>	<b>←</b>	183 Session progress		
	CPG ←	<b>←</b>	180 Ringing		
	Ringing tone				
	ANM <b>←</b>	<b>←</b>	200 OK INVITE		
		<b>→</b>	ACK		
	Cor	versation			
	<b>→</b>	<b>→</b>	BYE		
	RLC <del>(</del>	+	200 OK BYE		

TP304005	SIP reference: RFC 3261 [6]	FG	ISUP reference: 6 283 027 [1], clause 7.2.3.1.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG me		203 027 [1], clause 7.2.3.1.4
SIP selection	PICS 3/1 AND PICS 4/18		
criteria:	1 100 3/1 AND 1 100 4/10		
ISUP selection			
criteria:			
Test purpose:	ACM was sent after T <sub>I/W1</sub> expiry, after receipt	t of a 183 a	nd 180 contains a PSTN XML
	ProgressIndicator #7 a CPG(Progress) and a	CPG(Alerti	ing) are sent
	Ensure that the SUT, having sent the ACM me 183 Session progress message followed by a		
	containing the progress descriptions "Termina		
	sends two CPG messages respectively w "Alerting".	ith the <b>eve</b>	nt indicator set to "Progress" and
SIP Parameter	183 Session progress		
values:	180 Ringing		
ISUP Parameter	CPG 1; event indicator: Progress		
values:	CPG 2; event indicator: Alerting		
	BCI:		
	interworking indicator: no interworking enco	untered	
	ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access	ICDNI"	
Comments:	ISUP/BICC SUT	ISDIN	SIP
Comments.	IAM →		SIF
	T <sub>I/W1</sub> expiry		
	ACM(no indication) ←	<b>→</b>	INVITE
	CPG(Progress)	<b>←</b>	183 Session progress
	CPG(Alerting)	<b>←</b>	180 Ringing
	Ringing tone		5 0
	ANM <b>←</b>	<b>←</b>	200 OK INVITE
		<b>→</b>	ACK
	Conv	ersation	
	<b>→</b>	<b>→</b>	BYE
	RLC ←	+	200 OK BYE

TP304006	SIP reference	: RFC 3261 [6]	-	ISUP reference:
T00 (	IOLID OID /D : II/	0 1: (11 000		S 283 027 [1], clause 7.2.3.1.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message			
SIP selection	PICS 3/1 AND PICS 4	I/18		
criteria:				
ISUP selection				
criteria:	1011 / C			5ALXA# D
Test purpose:		* * * * *		N XML ProgressIndicator #7 and #x
	received, a CPG is se	nt contains an ATP with	PI #x	
				ter the expiry of Ti/W1, on receipt of
				containing the progress descriptions
	ATP:	SDN (#7) and PI_VALU	E the Pro	gressIndicator #7 is not sent in the
	AIF.			
	sends the CDG m	necessa with the avent i	ndicator	set to "progress" and the ATP
		t <b>or</b> set to PI_VALUE.	Iluicatoi	secto progress and the ATF
SIP Parameter	183 Session Progress			
values:		, , ndicator " <i>Terminating ac</i>	cess ISD	N"(#7)
1 4.1.4.5	PI VALUE	g as		(,,,,
ISUP Parameter	CPG; event indicator	r: progress		
values:	ATP progress indica	tor: PI_VALUE		
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>		
		T <sub>I/W1</sub> expiry		
	ACM(no indication)	<b>←</b>	<b>→</b>	INVITE
	CPG(Progress)	<b>←</b>	<b>←</b>	183 Session progress
	CPG(Alerting)	<b>←</b>	<b>←</b>	180 Ringing
		Ringing tone		
	ANM	<b>←</b> "	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conv	ersation	
		<b>→</b>	<b>→</b>	BYE
	RLC	<del>(</del>	<b>←</b>	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]	F	ISUP reference: S 283 027 [1], clause 7.2.3.1.4	
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message			
SIP selection	PICS 3/1 AND PICS 4/18	ooago		
criteria:				
ISUP selection				
criteria:				
Test purpose:	ACM was sent after T <sub>I/W1</sub> expiry, a 180 cover	ing a PST	N XML ProgressIndicator #7 and #x	
	received, a CPG is sent contains an ATP with	PI #x		
	Ensure that the SUT, having sent the ACM me			
	with PSTN XML ProgressIndicator "Termina	ting acces	s <i>ISDN"(#7)"</i> and " PI_VALUE ",	
	and a ODO man and the three areas to		A 4 - II A I - orbin - or II - or - I 4 I - ATD in - I - orbin - or	
	<ul> <li>sends a CPG message with the event inches</li> <li>the progress indicator set to " PI_VALU</li> </ul>		t to "Alerting" and the ATP including	
SIP Parameter	180 Ringing;	<u> </u>		
values:	PSTN XML ProgressIndicator: PI_VALUE			
ISUP Parameter	CPG; Event indicator: Alerting			
values:	ATP progress indicator: PI_VALUE			
Comments:	ISUP/BICC SUT		SIP	
	IAM →			
	T <sub>OIW1</sub> expiry	•		
	ACM(no indication) ←	<b>→</b>	INVITE	
	CPG(Alerting) ←	<b>←</b>	180 Ringing	
	Ringing tone			
	ANM	<b>←</b>	200 OK INVITE	
		<b>→</b>	ACK	
	Conv	ersation		
	<b>→</b>	<b>→</b>	BYE	
	RLC ←	<b>←</b>	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	ce: RFC 3261 [6]	F	ISUP reference: S 283 027 [1], clause 7.2.3.1.4	
TSS reference:	ISUP-SIP /Basic cal	I/ Sending of the CPG me		.o 200 021 [1], 014456 1.2.01114	
SIP selection	PICS 2/3 AND PICS		Jugo		
criteria:	1 100 2/0 / (14) 1 100	7 7/10			
ISUP selection	PICS 4/19				
criteria:	1.00 1,10				
Test purpose:	ACM was sent after	$T_{I/M/1}$ expiry, a 180 cover	ing a PS1	N XML ProgressIndicator #7 and #x	
		sent contains an ATP with		•	
	70001704, 4 07 0 10 0	on comanic arriver was			
	message containing "Interworking has or	Ensure that the SUT having sent the ACM message, on receipt of a 183 Call Progress message containing the <b>PSTN XML ProgressIndicator</b> " <i>Terminating access ISDN</i> "(#7) and "Interworking has occurred and has resulted in a telecommunication service change (#5)", the PSTN XML <b>BearerCapability</b> set to BC_VALUE:			
	the <b>BC</b> set to B and has resulte TMU_VALUE.	C_VALUE and the <b>progre</b>	ss indica	set to "Progress", the ATP containing ator set to "Interworking has occurred hange (#5)" and the TMU set to	
SIP Parameter	INVITE;				
values:		arer Capability: INVITE _			
	PSTN XML second	Bearer Capability: INVIT	E_BC2		
	400 O-II Dua F	OTNI VMI. Duo uuo o olu dii		and the second second by a	
	183 Call Progress; <b>PSTN XML ProgressIndicator</b> : Interworking has occurred and has				
	BC VALUE	resulted in a telecommunication service change (#5). PSTN XML BearerCapability:			
ISUP Parameter		udio3Kbit/s, G.711 A-law			
values:		Digital info T/A, G.711 A-I	aw		
10.000	TMR: 64 kbit/s prefe				
	TMR prime: Speech				
	CPG, event indicat				
	ATP BC: BC_VALU	E			
	ATP progress indic	cator: Interworking has oc	curred an	d has resulted in a telecommunication	
	service change (#5)				
	TMU: TMU_VALUE				
Comments:	ISUP/BICC	SUT		SIP	
	IAM	<b>→</b>			
		T <sub>OIW1</sub> expiry			
	ACM	<b>←</b>	<b>→</b>		
	CPG	<b>←</b>	<b>←</b>	183 Session progress	
	CPG	<b>←</b>	<b>←</b>	180 Ringing	
	l	Ringing tone	_		
	ANM	<b>←</b>	<del>(</del>	200 OK INVITE	
		_	→	ACK	
			ersation		
	D. 0	<del>)</del>	<b>→</b>	BYE	
	RLC	<b>+</b>		200 OK BYE	

TP304009	SIP refere	nce: 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 PIC				
ISUP selection criteria:	PICS 4/19				
Test purpose:	ACM was sent after	er T <sub>I/W1</sub> expiry, Fallback o	ccurs in the	e 180 Ringing	
	Ensure that the SUT in call having sent the ACM message, on receipt of an 180 Ringing message containing the <b>PSTN XML BearerCapability</b> set to BC_VALUE and the <b>PSTN XML ProgressIndicator</b> set to " <i>Terminating access ISDN"(#7) and</i> "Interworking has occurred and has resulted in a telecommunication service change (#5)":				
	the <b>BC</b> set to	BC_VALUE and the <b>prog</b> ited in a telecommunication	ess indica	set to "Alerting", the <b>ATP</b> containing ator set to "Interworking has occurred hange (#5)" and the <b>TMU</b> set to	
SIP Parameter values:		INVITE; PSTN XML first Bearer Capability: INVITE _BC1 PSTN XML second Bearer Capability: INVITE _BC2			
	PSTN XML Progre occurred and has	PSTN XML ProgressIndicator: "Terminating access ISDN"(#7) and Interworking has occurred and has resulted in a telecommunication service change (#5) PSTN XML BearerCapability: BC_VALUE			
ISUP Parameter		IAM; <b>USI</b> : Speech/audio3Kbit/s, G.711 A-law			
values:	<b>USI</b> prime: Unrest	USI prime: Unrestr. Digital info T/A, G.711 A-law			
		TMR: 64 kbit/s preferred			
	TMR prime: Speech/audio3Kbit/s				
	CDC: event indicator: Alerting				
	CPG; event indicator: Alerting ATP BC: BC_VALUE				
			ccurred ar	nd has resulted in a telecommunication	
	service change (#8	5)			
	TMU: TMU_VALU				
Comments:	ISUP/BICC	SUT		SIP	
	IAM	<b>→</b>			
		T <sub>OIW1</sub> expir	У		
	ACM	<b>←</b>	<b>→</b>	INVITE	
	CPG	<b>←</b> Ringing tone	<b>←</b>	180 Ringing	
	ANM	E tone	<b>←</b> →	200 OK INVITE ACK	
		Con	versation	-	
		<b>→</b>	<b>→</b>	BYE	
	RLC	<b>←</b>	+	200 OK BYE	

TP304010	SIP refer	ence: RFC 3261 [6]		ISUP reference: S 283 027 [1], clause 7.2.3.1.4.1		
TSS reference:	ISLIP-SIP /Rasic	call/ Sending of the CPG me		5 203 027 [1], clause 7.2.3.1.4.1		
SIP selection	PICS 2/3 AND PI	ICS 4/18	ssage			
criteria:	1100 20 7110 4710					
ISUP selection	PICS 4/19					
criteria:						
Test purpose:	ACM was sent at	fter T <sub>I/W1</sub> expiry, Fallback o	curs in the	e 183 Session Progress		
	Ensure that the SUT having sent the ACM message, on receipt of a 183 Session Progress message containing the <b>BC</b> SET to BC_VALUE and the <b>PSTN XML ProgressIndicator</b> set to " <i>Terminating access ISDN"(#7)</i> , "Interworking has occurred and has resulted in a telecommunication service change (#5)" and "In-band information or appropriate pattern is now available (#8)":					
	appropriate p	pattern is now available", the	ATP conting has oc	set to "In-band information or taining the <b>BC</b> set to BC_VALUE and securred and has resulted in a <b>MU</b> set to TMU_VALUE.		
SIP Parameter	INVITE;					
values:		Bearer Capability: INVITE				
		ond Bearer Capability: INVI	TE _BC2			
	183 Session Prog PSTN XML	gress;				
	-	or: Interworking has occurre	d and has	resulted in a telecommunication		
	service change (#	ProgressIndicator: Interworking has occurred and has resulted in a telecommunication				
	ProgressIndicat	ProgressIndicator: In-band information or appropriate pattern is now available (#8) PSTN XML BeaererCapability: BC_VALUE				
ISUP Parameter	IAM; USI: Speech/audio3Kbit/s, G.711 A-law					
values:	<b>USI</b> prime: Unrestr. Digital info T/A, G.711 A-law					
		TMR: 64 kbit/s preferred				
	TMR prime: Speech/audio3Kbit/s					
	0004	CPG 1; event indicator: In-band information or appropriate pattern is now available				
	ATP BC: BC_VA		or approp	riate pattern is now available		
			ccurred an	d has resulted in a telecommunication		
	service change (#		ocurrou ar	a nac recance in a telecommunication		
	TMU: TMU_VALL					
Comments:	ISUP/BICC	SUT		SIP		
	IAM	<b>→</b>				
		T <sub>OIW1</sub> expir	/			
	ACM	<b>←</b>	<b>→</b>	INVITE		
	CPG 1	<b>←</b>	<b>←</b>	183 Session progress		
	CPG	<b>←</b>	<b>←</b>	180 Ringing		
		Ringing tone				
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE		
			<b>→</b>	ACK		
			ersation/			
		<b>→</b>	<b>→</b>	BYE		
	RLC	+	+	200 OK BYE		

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

TP304011	SIP reference: RFC 3261 [6]		ISUP reference:	
T00 (	10110 010 /0 : 11/0 1: (11 000		S 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 <sub>I/W1</sub> expiry, PSTN XML HLC received in a 183 mapping in the ATP			
	contained in the CPG		77 3	
	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 ch	nange (#5)".	
SIP Parameter	183 Session Progress;			
values:	<b>PSTN XML ProgressIndicator</b> : Interworking telecommunication service change (#5) and "	Terminatiı	ng access ISDN"(#7)	
	PSTN XML HighLayerCompatibility: HLC_VALUE2 (PIXIT)			
ISUP Parameter	CPG 1, Event indicator: Progress			
values:	ATP HLC: HLC_VALUE2 (PIXIT)		al la a constitue d'in contact a con	
	ATP progress indicator: Interworking has occurred and has resulted in a telecommunication			
0	service change (#5)		OID	
Comments:	ISUP/BICC SUT		SIP	
	IAM →			
	T <sub>OIW1</sub> expiry			
	ACM ←	<b>→</b>		
	CPG 1 ←	<b>←</b>	183 Session progress	
	CPG ←	<b>←</b>	180 Ringing	
	Ringing tone			
	ANM <b>←</b>	<b>←</b>	200 OK INVITE	
		<b>→</b>	ACK	
		ersation		
	<b>→</b>	<b>→</b>	BYE	
	RLC ←	+	200 OK BYE	

TP304012	SIP reference: RFC 3261 [6]		ISUP reference:	
TCC reference	IOUD OID /Darie and // Or or dispersed the ODO or		S 283 027 [1], clause 7.2.3.1.4	
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG m	essage		
SIP selection	PICS 4/18			
criteria:				
ISUP selection				
criteria:	1014	0	·	
Test purpose:	ACM was sent after $T_{I/W1}$ expiry, PSTN XML	. HLC rece	ived in a 180 mapping in the ATP	
	contained in the CPG			
	Ensure that the SUT, having sent the ACM n			
	with PSTN XML ProgressIndicator set to "Int			
	telecommunication service change (#5)" "Te		ccess ISDN"(#7)and with a PSTN XML	
	HighLayerCompatibility set to HLC_VALUE	::		
	and the CDC masses with as a state	!aata= ==+:	to II A loutill the ATD in shortlings the LUC	
		<ul> <li>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</li> </ul>		
SIP Parameter	has resulted in a telecommunication ser	vice change	e (#5) .	
	180 Ringing;	. haa aaau	rad and has requited in a	
values:	PSTN XML ProgressIndicator: Interworking has occurred and has resulted in a			
	telecommunication service change (#5), "Terminating access ISDN"(#7)  PSTN XML HighLayerCompatibility: HLC_VALUE2 (PIXIT)			
ISUP Parameter	CPG, Event indicator: Alerting			
values:	ATP HLC: HLC_VALUE2 (PIXIT)			
values.	ATP progress indicator: Interworking has o	ccurred an	d has resulted in a telecommunication	
	service change (#5)	oourrou urr	a nao robattoa in a tolobominambatton	
Comments:	ISUP/BICC SUT		SIP	
30	IAM →		<del></del>	
	T <sub>OIW1</sub> expir	V		
	J		INIVITE	
	ACM ←	<b>→</b>	INVITE	
	CPG ←	+	180 Ringing	
	Ringing tone	_	000 01/ 10 1/ 175	
	ANM ←	<del>(</del>	200 OK INVITE	
	_	→	ACK	
		versation	5.75	
	<b>→</b>	<b>→</b>	BYE	
	RLC ←	<u>+</u>	200 OK BYE	

TP304013	SIP reference: RFC 3261 [6]	F	ISUP reference: S 283 027 [1], clause 7.2.3.1.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG me		.o 200 027 [1], ciduse 7.2.5.1.4
SIP selection	Poor on /Basic cail/ contains of the or o me	Jooage	
criteria:			
ISUP selection	PICS 4/9		
criteria:			
Test purpose:	ACM sent after INVITE was sent, 183 receive ProgressIndicator #7 mapped into BCI in the		
	Ensure that the SUT, having sent automatica Ringing message followed by a 183 Session <b>ProgressIndicator</b> " <i>Terminating access ISD</i> .	Progress i N"(#7):	message with <b>PSTN XML</b>
	<ul> <li>sends two CPG message respectively wi "Progress".</li> </ul>	th the eve	ent indicator set to "Alerting" and
SIP Parameter	180 Ringing;		
values:	183 Session Progress;		
ISUP Parameter	CPG 1; event indicator: Alerting		
values:	ISUP indicator: ISUP is used all the way		
	ISDN access indicator: ISDN		
	Interworking indicator: Interworking not enountered		
	CPG 2; event indicator: Progress		
	ISUP indicator: ISUP is used all the way		
	ISDN access indicator: ISDN		
	Interworking indicator: Interworking not enour	ntered	
Comments:	ISUP/BICC SUT		SIP
	IAM →		
	ACM ←	<b>→</b>	INVITE
	CPG 1 ←	<b>←</b>	180 Ringing
	CPG 2 ←	<b>←</b>	183 Session progress
	Ringing tone		
	ANM ←	<b>←</b>	200 OK INVITE
		<b>→</b>	ACK
		ersation/	
	REL →	<b>→</b>	BYE
	RLC +	+	200 OK BYE

TP304014	SIP reference: RFC 3261 [6]	E:	ISUP reference: S 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:	<ul> <li>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</li> <li>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:</li> <li>sends a CPG message with the event indicator set to "Alerting" and the ATP including the progress indicator set to PI_VALUE.</li> </ul>		
SIP Parameter values:	180 Ringing; progress indicator: PI_VALUE		
ISUP Parameter	CPG; Event indicator: Alerting		
values:	ATP progress indicator: PI_VALUE		
Comments:	SUT		SIP
	IAM →		
	ACM ←	<b>→</b>	INVITE
	CPG ←	<b>←</b>	180 Ringing
	Ringing tone		
	ANM ←	<b>←</b>	200 OK INVITE
	Conv	→ ersation	ACK
	REL →	<b>→</b>	BYE
	RLC ←	<b>←</b>	200 OK BYE

Values and additional selection criteria for test purposes TP304014			
VA_01	PI_VALUE = Call is not end-to-end ISDN (#1)		
VA 02	PL_VALUE = Destination address is non-ISDN (#2)		

TP304015	SIP referer	nce: RFC 3261 [6]	E	ISUP reference: S 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	PICS 4/9			
criteria:				
Test purpose:	mapped into OBCI Ensure that the SU	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 <b>event indicator</b> set to "Alerting" and <b>OBCI in-band</b> information set to "yes".			
SIP Parameter	180 Ringing; PSTN	I XML ProgressIndicator "In	-band info	ormation or appropriate pattern is now
values:	available" (#8)			
ISUP Parameter	IAM: USI: USI_VAL			
values:	CPG; Event indica			
	OBCI in-band: yes			
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>		
	ACM	<b>←</b>	<b>→</b>	INVITE
	CPG	←	<b>←</b>	180 Ringing
		Ringing tone	-	
	ANM	<b>*</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conv	ersation	
		<b>→</b>	<b>→</b>	BYE
	RLC	+	<b>←</b>	200 OK BYE

TP304016	SIP reference: F	RFC 3261 [6]		ISUP reference:
				S 283 027 [1], clause 7.2.3.2.6
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message			
SIP selection				
criteria:				
ISUP selection	PICS 4/9			
criteria:				
Test purpose:	ACM sent after INVITE was sent, 180 received contains a P-Early-Media header mapped into the OBCI "inband info available"  Ensure that the SUT, having received an IAM with the <b>USI field</b> 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 <b>event indicator</b> set to "Alerting" and <b>OBCI in-band</b> information set to "yes".			
SIP Parameter	180 Ringing; P-Early-Me	dia header authorizing	g early m	edia
values:		, ,		
ISUP Parameter	IAM: USI: USI_VALUE;			
values:	CPG; Event indicator: Alerting			
	OBCI in-band: yes			
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>		
	ACM	<del>(</del>	<b>→</b>	INVITE
	CPG	<b>←</b>	<b>←</b>	180 Ringing
	Rin	iging tone		
	ANM	<del>(</del>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conv	ersation	
		<b>→</b>	<b>→</b>	BYE
	RLC	<del>(</del>	<b>←</b>	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:
		E	S 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 rece	eived	
	Ensure that the SUT having sent the ACM me	essage,	on receipt of a 200 OK INVITE for this
	call, it shall stop timer TOIW2 (if running):		
	0 14444 14 1 14 15 14 15 16 16 16 16		
	Send ANM as determined by BICC/ISUP	•	
	<ul> <li>Stop any existing awaiting answer indicat</li> </ul>	ion (e.g	i. ringing tone).
SIP Parameter	200 OK INVITE;		
values:			
ISUP Parameter	ANM;		
values:			
Comments:	ISUP/BICC SUT		SIP
	IAM →	<b>→</b>	INVITE
	ACM ←	<b>←</b>	180 Ringing
	Ringing tone		
	ANM ←	<b>←</b>	200 OK INVITE
		<b>→</b>	ACK
	Conve	rsation	
	REL →	<b>→</b>	BYE
	RLC ←	<b>←</b>	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	PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:	An ANM is sent after a 200 OK INVITE mapped into the ATP in the ANM  Ensure that the SUT, having sent the ACM		Ç
	containing the PSTN XML ProgressIndica	or set to	PI_VALUE:
	<ul> <li>sends the ANM message with the ATP set to PI_VALUE.</li> </ul>	including	the PSTN XML ProgressIndicator
SIP Parameter values:	200 OK; PSTN XML ProgressIndicator: P	_VALUE	(PIXIT)
ISUP Parameter values:	ANM; ATP Progress Indicator: PI_VALUE	(PIXIT)	
Comments:	ISUP/BICC SUT		SIP
	IAM →	<b>→</b>	INVITE
	ACM ←	<b>←</b>	180 Ringing
	Ringing tone		3 3
	ANM <b>←</b>	<b>←</b>	200 OK INVITE
		<b>→</b>	ACK
	Con	ersation	
	REL →	<b>→</b>	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	PICS 4/18		
criteria:			
ISUP selection			
criteria:			
Test purpose:		received. PSTN XML LowLayerCompatibility	
	mapped into the ATP in the ANM		
	Ensure that the SUT, having sent the ACM monotoning the <b>PSTN XML LowLayerCompate</b>		
	Containing the PSTN AML LOWLayer Compan	aubility set to LLC_VALUE.	
	<ul> <li>sends the ANM message with the ATP in</li> </ul>	ncluding the LLC set to LLC VALUE	
SIP Parameter	200 OK; PSTN XML LowLayerCompatibility		
values:	200 ON, FORM ANIL LOWLAYER COMPANDING. LLO_VALUE (FIAIT)		
ISUP Parameter	ANM; ATP LLC: PI_VALUE (PIXIT)		
values:	/ www. All EEG. I i_ v/lege (i b/li)		
Comments:	ISUP/BICC SUT	SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	Ringing tone	3 3	
	ANM <b>←</b>	◆ 200 OK INVITE	
		→ ACK	
	Conve	ersation	
	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 M	lessage	e (ANM)/
SIP selection	PICS 4/18 AND PICS 2/3		
criteria:			
ISUP selection	PICS 4/19		
criteria:			
Test purpose:	An ANM is sent after a 200 OK INVITE is i	eceive	d. PSTN XML HighLayerCompatibility
	mapped into the ATP in the ANM		an receipt of a 200 OK Massacra
	Ensure that the SUT, having sent the ACM me containing the PSTN XML HighLayerCompate		
	Containing the F3TN AME HighLayerCompa	ibility	Set to FIEC_VALUE 1.
	sends the ANM message with the ATP in	cluding	the <b>HLC</b> set to HLC_VALUE1.
SIP Parameter	200 OK; PSTN XML HighLayerCompatibility		
values:			,
ISUP Parameter	IAM; ATP HLC1: HLC_VALUE1 (PIXIT)		
values:	ATP HLC2: HLC_VALUE2 (PIXIT)		
	AND ATRICO LICO VALUES (DIVIT)		
	ANM; ATP HLC: HLC_VALUE2 (PIXIT)		OLD
Comments:	ISUP/BICC SUT	_	SIP
	IAM →	<b>→</b>	INVITE
	ACM ←	<b>←</b>	180 Ringing
	Ringing tone		
	ANM ←	<del>-</del>	200 OK INVITE
		<b>→</b>	ACK
	Conver	sation	
	REL →	<b>→</b>	BYE
	RLC <b>←</b>	+	200 OK BYE

TP305005	SIP refere	nce: 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 PI	CS 2/3		
ISUP selection criteria:	PICS 4/19			
Test purpose:	mapped into the A	TP and TMU in the AN	IM	ived. PSTN XML BearerCompatibility
	sent the ACM mes	JT, having received the sage, on receipt of a 2 yet to BC_VALUE:		age indicating BC fallback and having sage with <b>PSTN XML</b>
	<ul> <li>sends the AN TMU set to TM</li> </ul>		<b>TP</b> including	the <b>BC</b> set to BC_VALUE and the
SIP Parameter	INVITE;			
values:		earer Capability: INV		
	200 OK:	PSTN XML second Bearer Capability: INVITE _BC2		
	PSTN XML BeaererCapability: BC_VALUE			
ISUP Parameter		/audio3Kbit/s, G.711 A		
values:	USI prime: Unrestr. Digital info T/A, CLEARMODE			
	TMR: 64 kbit/s preferred			
	TMR prime: Speech/audio3Kbit/s			
	ANM; ATP BC: BC	: VALUE		
	TMU: TMU_VALU	_		
Comments:	ISUP/BICC	SU		SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
		Ringing tone	-	
	ANM	<b>←</b>	<del>(</del>	200 OK INVITE
		_	→ onversation	ACK
	REL	→	onversation	BYE
	RLC	É	<b>+</b>	200 OK BYE

Values and additional selection criteria for test purposes TP TP305005			
VA_01	TMU_VALUE: speech	BC_VALUE: speech	
	ISUP_VALUE: UDI/TA		
VA 02	TMU_VALUE: 3.1 kHz	BC_VALUE: 3.1 kHz	

TP305006	SIP reference: RFC	3261 [6]	ı	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	Ī		
SIP Parameter	INVITE;			
values:	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 USI prime: Unrestr. Digital inf TMR: 64 kbit/s preferred TMR prime: Speech/audio3K ANM; ATP BC: USI_VALUE TMU: TMR_VALUE	to T/A, CLEARMO	ODE	
Comments:	ISUP/BICC	SUT		SIP
	IAM →		<b>→</b>	INVITE
	ACM ← Ringing to		<b>←</b>	180 Ringing
	ANM		<b>←</b> →	200 OK INVITE ACK
		Conver	rsation	
	REL →		<b>→</b>	BYE
	RLC +		+	200 OK BYE

Values and additional selection criteria for test purposes TP305006					
VA_01	TMU_VALUE: speech				
	USI_VALUE: speech				
VA_02	TMU_VALUE: 3,1 kHz audio				
	USI_VALUE: 3,1 kHz audio				

TP305007	SIP reference: RFC 3261 [6]	E	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 rece ProgressIndicator #5, mapped into the ATP in				
	Ensure that the SUT, having sent the ACM message, on receipt of a 200 OK Message containing the <b>PSTN XML HighLayerCompatibility</b> set to HLC_VALUE1 and the <b>PSTN XML ProgressIndicator</b> set to "Interworking has occurred and has resulted in a telecommunication service change (#5)":				
	<ul> <li>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)".</li> </ul>				
SIP Parameter	200 OK INVITE				
values:	PSTN XML HighLayerCompatibility: HLC_V				
	PSTN XML ProgressIndicator: Interworking has occurred and has resulted in a				
	telecommunication service change (#5)				
ISUP Parameter	IAM; ATP HLC1: HLC_VALUE1 (PIXIT)				
values:	ATP HLC2: HLC_VALUE2 (PIXIT)				
	ANM; ATP HLC: HLC_VALUE1 (PIXIT)				
	ATP Progress Indicator: Interworking has oc	currea	and has resulted in a		
Comments:	telecommunication service change (#5) ISUP/BICC SUT		SIP		
Comments:			•		
	IAM →	<b>→</b>	INVITE		
	ACM ←	+	180 Ringing		
	Ringing tone	_	200 OK INIVITE		
	ANM ←	<del>(</del>	200 OK INVITE		
	_	<b>→</b>	ACK		
	Conver	_	5)/5		
	REL →	<b>→</b>	BYE		
	RLC <b>←</b>	<u> </u>	200 OK BYE		

TP305008	SIP refere	ence: 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			
	sent the ACM me	ssage, on receipt of a 20	0 OK Mes	age indicating BC fallback and having sage with <b>PSTN XML</b> LUE and the <b>PSTN XML</b>
		or set to "Interworking han service change (#5)":	s occurred	d and has resulted in a
	TMU set to T		gress Ind	the <b>BC</b> set to BC_VALUE and the <b>icator</b> set to "Interworking has on service change (#5)".
SIP Parameter	INVITE:			5 \ /
values:	PSTN XML first E	Bearer Capability: INVIT	E BC1	
		nd Bearer Capability: IN		2
	200 OK:			
	PSTN XML BearerCapability: BC_VALUE			
	PSTN XML ProgressIndicator: Interworking has occurred and has resulted in a			
	telecommunication service change (#5)			
ISUP Parameter	IAM; USI: Speech/audio3Kbit/s, G.711 A-law			
values:	USI prime: Unrestr. Digital info T/A, CLEARMODE			
	TMR: 64 kbit/s preferred			
	TMR prime: Spee	ech/audio3Kbit/s		
	ANM; <b>ATP BC</b> : B			
		dicator: Interworking has	coccurred	and has resulted in a
	telecommunication service change (#5)			
0	TMU: TMU_VALU			OID
Comments:	ISUP/BICC	SUT	_	SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	+	180 Ringing
		Ringing tone		
	ANM	<b>←</b>	<del>(</del>	200 OK INVITE
			<b>→</b>	ACK
			nversation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	<b>+</b>	+	200 OK BYE

Values and additional selection criteria for test purposes TP305008			
VA_01	TMU_VALUE: speech	BC_VALUE: speech	
VA 02	TMU VALUE: 3,1 kHz	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	NOT PICS 4/9		
criteria:			
Test purpose:	CON is sent after 200 was received		
	Ensure that the SUT, having not sent the ACN		
	for this call, it shall stop timer TOIW2 (if runnir	ng):	
	0 10011 1 1 5:00 (0:15		
	Send CON as determined by BICC/ISUP		
	Stop any existing awaiting answer indicat	tion (e.g. ringing tone) BCI encoded as	
	followed:		
	Interworking indicator: interworking		
	ISUP indicator: ISUP not used all the way		
OID Damana dan	ISDN access indicator: terminating	access non-ISDN	
SIP Parameter	200 OK INVITE;		
values:			
ISUP Parameter	CON: Interworking indicator: interworking encountered		
values:	ISUP indicator: ISUP not used all the way ISDN access indicator: terminating access non-ISDN		
Comments:	ISUP/BICC SUT	SIP	
Comments.	IAM →	→ INVITE	
	1		
	CON ←	€ 200 OK INVITE	
	_	→ ACK	
	Conve		
	REL →	→ BYE	
	RLC <b>←</b>	← 200 OK BYE	

TP306002	SIP referer	nce: 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 ANI	D PICS 4/17		
Test purpose:	IAM received with	TMR 64 kBit/s. BCI in t	ne CON ind	dicates ISDN access
		IT, having not sent the a stop timer TOIW2 (if ru		age, on receipt of a 200 OK INVITE
	<ul> <li>Send CON as</li> </ul>	determined by BICC/IS	UP proced	lures.
		-	•	g. ringing tone) BCI encoded as
		rking indicator: no inte		encountered (0)
	ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access ISDN"			ess ISDN"
SIP Parameter values:	200 OK INVITE			
ISUP Parameter	CON:			
values:		rking indicator: no inte	-	encountered (0)
		dicator: ISUP used all		IODNII
Comments:	ISUP/BICC	cess indicator: "termi SUT	nating acce	SIP
Comments.	IAM	→	<b>→</b>	INVITE
	CON	É	<i>+</i>	<u>-</u>
		•	÷	ACK
		Со	nversation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	<b>←</b>	+	200 OK BYE

TP306003	SIP reference: I	RFC 3261 [6]	ı	ISUP reference: ES 283 027 [1], clause 7.2.3.1.5
TSS reference:	ISUP-SIP/Basic call/ Se	nding of the Connect I	Messag	je (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	<b>→</b>	<b>→</b>	INVITE
	CON	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conver	rsation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	+	<b>←</b>	200 OK BYE

TP306004	SIP reference: R	FC 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	PICS 4/18			
criteria:				
ISUP selection				
criteria:				
Test purpose:	A CON is sent after a 2 mapped into the ATP in the		eceive	d. PSTN XML LowLayerCompatibility
	mapped into the ATT in t	ne oon		
	Ensure that on receipt of	a 200 OK Message o	ontain	ing the PSTN XML
	LowLayerCompatibility	•		
	sends the CON message			
SIP Parameter	200 OK INVITE: PSTN XML LowLayerCompatibility: LLC_VALUE (PIXIT)			
values:				
ISUP Parameter	CON: <b>ATP LLC</b> : PI_VALI	JE (PIXIT)		
values:				
Comments:	ISUP/BICC	SUT	_	SIP
	IAM	<b>→</b>	<b>→</b>	
	CON	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conver	sation	
	REL	<b>→</b>	<b>→</b>	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 HighLayer			
ISUP Parameter values:	IAM; ATP HLC1: HLC_VALUE1 (PIXIT) ATP HLC2: HLC_VALUE2 (PIXIT) ACON: ATP HLC: HLC_VALUE2 (PIXIT)	1		
Comments:	ISUP/BICC SU	Г	SIP	
	IAM →	<b>→</b>	INVITE	
	CON ←	<b>←</b>	200 OK INVITE	
		<b>→</b>	ACK	
	C	onversation		
	REL →	<b>→</b>	BYE	
	RLC ←	+	200 OK BYE	

TP306006	SIP reference: RFC 3261 [6]		ISUP reference:	
11 300000	Sir reference. Ki C 3201 [0]		ES 283 027 [1], clause 7.2.3.1.5	
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/			
SIP selection	PICS 4/18 AND PICS 2/3			
criteria:				
ISUP selection	PICS 4/19			
criteria:				
Test purpose:	A CON is sent after a 200 OK INVITE is into the ATP and TMU in the CON	received. P	STN XML BearerCompatibility mapped	
	Ensure that the SUT, having received th a 200 OK Message with <b>PSTN XML Bea</b>			
	sends the CON message with the <b>ATP</b> is set to TMU_VALUE.	ncluding the	BC set to BC_VALUE and the TMU	
SIP Parameter	INVITE;			
values:	PSTN XML first Bearer Capability: INV			
	PSTN XML second Bearer Capability:	INVITE _BC	·Z	
	200 OK INVITE			
	PSTN XML BeaererCapability: BC_VALUE			
ISUP Parameter	IAM; USI: Speech/audio3Kbit/s, G.711 A-law			
values:	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 SU	Т	SIP	
	IAM →	<b>→</b>	INVITE	
	CON ←	<b>←</b>	200 OK INVITE	
		<b>→</b>	ACK	
		onversation		
	REL →	<b>→</b>	BYE	
	RLC <b>←</b>	+	200 OK BYE	

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

TP306007	SIP reference: RF	C 3261 [6]		ISUP reference: ES 283 027 [1], clause 7.2.3.1.5
TSS reference:	ISLID SID/Rasic call/ Sandi	ng of the Connect I		:
SIP selection	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/ PICS 4/18 AND PICS 2/3			
criteria:	FICS 4/ 18 AND FICS 2/3			
ISUP selection	PICS 4/19			
criteria:	1.100 17.10			
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 v	vith the <b>ATP</b> includi	na the	BC set to USL VALUE and the TMU
	sends the CON message with the <b>ATP</b> including the <b>BC</b> set to USI_VALUE and the TMU set to TMU VALUE.			
SIP Parameter	INVITE:			
values:	PSTN XML first Bearer Capability: INVITE _BC1			
	PSTN XML second Bearer Capability: INVITE _BC2			
IOUR D	200 OK INVITE: no BC	1 ::/ 0 744 4 1		
ISUP Parameter values:	IAM; USI: Speech/audio3Kbit/s, G.711 A-law			
values:	USI prime: Unrestr. Digital info T/A, G.711 A-law TMR: 64 kbit/s preferred			
	TMR prime: Speech/audio	3Khit/s		
	Pillio. Specciliadaic	OI (DIQ O		
	CON: ATP BC: USI_VALUE			
	TMU: TMU_VALUE			
Comments:	ISUP/BICC	SUT		SIP
		<b>→</b>	<b>→</b>	INVITE
	CON	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
	DEL	Conver		DVE
	REL	<b>→</b>	<b>→</b>	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	PICS 4/18 AND PICS 2/3		
criteria:			
ISUP selection	PICS 4/19		
criteria:			
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_VALUE		
	to "Interworking has occurred and has resulted		
	(#5)"		ŭ
	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		
SIP Parameter	telecommunication service change (#5)".		
values:	200 OK INVITE   PSTN XML HighLayerCompatibility: HLC_VALUE1 (PIXIT)		
values.	PSTN XML ProgressIndicator: Interworking has occurred and has resulted in a		
	telecommunication service change (#5)		
ISUP Parameter	IAM; ATP HLC1: HLC_VALUE1 (PIXIT)		
values:	ATP HLC2: HLC_VALUE2 (PIXIT)		
	CON: ATP HLC: HLC_VALUE1 (PIXIT)		
	ATP Progress Indicator: Interworking has occurred and has resulted in a		
0	telecommunication service change (#5)		
Comments:	ISUP/BICC SUT		SIP
	IAM →	<b>→</b>	INVITE
	CON ←	<b>←</b>	200 OK INVITE
	Conve	-	ACK
	Conversation		
	REL → RLC ←	→ ←	BYE 200 OK BYE
	INLO T	7-	ZUU UN DIE

TP306009	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		
SIP selection criteria:	PICS 4/18 AND PICS 2/3	<b>J</b> - (	
ISUP selection criteria:	PICS 4/19		
Test purpose:	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		
	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 includes set to TMU_VALUE and the Progress Indicates resulted in a telecommunication service chan	tor set to "Interworking has occurred and has	
SIP Parameter	INVITE;		
values:	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)		
ISUP Parameter	IAM; USI: Speech/audio3Kbit/s, G.711 A-law		
values:	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		
Comments:	ISUP/BICC SUT	SIP	
	IAM →	→ INVITE	
	CON ←	← 200 OK INVITE  → ACK	
	Convo	rsation	
	REL →	→ BYE	
	RLC +	€ 200 OK BYE	
l	1.1.20	- ZOO ON DIE	

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

## 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 be receipt of a REL message:  no action is required on the SIP side other are in progress.	er than to terminate local procedures if any	
SIP Parameter			
values:	DEL OV TOUR (DIVIT)		
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT)		
	location: LOC_ISUP (PIXIT)	CID	
Comments:	ISUP/BICC SUT	SIP	
	REL →		
	RLC <del>←</del>		

TP307002	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.2.3.1.8
TSS reference:	ISLID SID/Pagia call/ Pagaint of the Palagae n	
SIP selection	ISUP-SIP/Basic call/ Receipt of the Release n	nessage (REL)/
criteria:		
ISUP selection		
criteria:		
	DEL received DVE is contacted ACK for 200	OK was southefore south dislocus
Test purpose:	REL received, BYE is sent after ACK for 200	OK was sent before early dialogue
	received.  The SUT shall send a BYE request. The CV_ISUP shall be mapped to the Reason	message <b>before</b> a 200 OK response (any) s a confirmed dialogue: til a SIP 200 OK INVITE response has been cause Value Indicator parameter defined as
SIP Parameter	cause value: CV_SIP (PIXIT)	
values:		
ISUP Parameter	REL: cause value: CV_ISUP (PIXIT)	
values:	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]	ES	ISUP reference: 283 027 [1], clause 7.2.3.1.8
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Relea	se message (I	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 IA sending an INVITE message. On receipt of		
	message has been received:		
	The SUT shall hold the REL message was been received.	e. A CANCEL	is sent when any SIP response
	On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a <b>BYE</b> 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	BYE: cause value: CV_SIP (PIXIT)		
values:			
ISUP Parameter	REL: cause value: CV_ISUP (PIXIT)		
values:	location: LOC_ISUP (PIXIT)		
Comments:	ISUP/BICC SU	ΙΤ	SIP
	IAM →	<b>→</b>	INVITE
		<b>←</b>	100 Trying
	REL →		
	RLC ←		
		<b>→</b>	CANCEL
		<b>←</b>	200 OK INVITE
		<b>←</b>	200 OK CANCEL
		<b>→</b>	ACK
		<b>→</b>	BYE
		+	200 OK BYE

TP307004	SIP reference: RFC 3261 [6]	ES	ISUP reference: 283 027 [1], clause 7.2.3.1.8
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Releas	e message (F	REL)/
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	<ul> <li>REL received before early dialogue is estated.</li> <li>Ensure that the SUT after receiving the IAI sending an INVITE message. On receipt of the message defined as SIP_MESSAGE in the SUT shall hold the REL message received.</li> <li>The SUT shall send a CANCEL requedefined as CV_ISUP shall be mapped.</li> </ul>	If with the core is a REL mess as been esta until a <b>SIP_N</b> st. The cause	age before an early dialogue with blished:  MESSAGE_VA response has been a Value Indicator parameter
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 SU	Γ	SIP
	IAM REL RLC  ←	<b>→</b>	INVITE
		<b>←</b>	SIP MESSAGE VA
		<b>→</b>	= =
		<b>←</b>	200 OK CANCEL
		<b>←</b>	487 Request terminated
		<b>→</b>	ACK

TP307005	SIP reference: RFC 3261 [6]	ES	ISUP reference: 283 027 [1], clause 7.2.3.1.8
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Rele	ase message (F	REL)/
SIP selection			
criteria:			
ISUP selection criteria:			
Test purpose:	REL received, BYE is sent after ACK for	200 OK was se	ent in early dialogue
	Ensure that the SUT after receiving the I sending an INVITE message. On receipt message has been received:  • The SUT shall send a BYE request Indicator parameter defined as CV_SIP.	of a REL mess	age <b>after</b> a 200 OK response as been sent. The cause Value
SIP Parameter	BYE: cause value: CV_SIP (PIXIT)		
values: ISUP Parameter	DEL COURS Value CV ISUD (DIVIT)		
values:	REL: cause value: CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)		
Comments:	` '	UT	SIP
Johnnients.	IAM →	••	INVITE
	ACM ←	<b>É</b>	180 Ringing
	ANM ←	÷	200 OK INVITE
	REL →	_	
	RLC •	<b>→</b>	ACK
		<b>→</b>	BYE
		<b>←</b>	200 OK BYE

TP307006	SIP reference: RFC 3261 [6]	ES 28	ISUP reference: 3 027 [1], clause 7.2.3.1.8
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release r		
SIP selection criteria:	, , , , , , , , , , , , , , , , , , , ,	<u>-</u>	F
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 →	<b>→</b>	INVITE
		<b>←</b>	SIP_MESSAGE_VA
	REL →		
	RLC		
			CANCEL
		· ·	200 OK CANCEL
			487 Request terminated ACK

#### Table 11

Values for test purpose TP307004; TP307006		
VA	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 Recommendation 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 C	ause Value as defined in	ITU-T Recommendation 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.

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

TP308001	SIP reference: RFC 326	1 [6]		ISUP reference:
				ES 283 027 [1], clause 7.2.3.1.7
TSS reference:	ISUP-SIP /Basic call/ Sending of	f the Release m	essa	ge (REL)/
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	BYE received, REL cause #16 is	s sent		
	of a BYE message where a Rea Cause Value is <b>not</b> included:	son header field	d with	ut an INVITE message and on receipt ITU-T Recommendation Q.850 [5]
SIP Parameter	9			, , ,
values:				
ISUP Parameter	REL; Cause value "Normal call of	clearing"		
values:		· ·		
Comments:	ISUP/BICC	SUT		SIP
	IAM →		<b>→</b>	INVITE
	ACM ←		<b>←</b>	180 Ringing
	Ringing tone	9		
	ANM 🗲		<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conversa	ation	-
	REL ←	301110100	<b>←</b>	BYE
	RLC →		÷	200 OK BYE
<u></u>	1			

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 Re		
SIP selection criteria:	PICS 4/11		.ge (\(\frac{1}{2}\)
ISUP selection criteria:			
Test purpose:	BYE Reason header #x received, REL of	ause #x is s	ent
	Ensure that the SUT after receiving the of a BYE message where a Reason hea Cause Value is included:  sends a REL message. The Cause ISUP Cause Value field in the ISUF	der field with Value is in t	
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 ←	<b>←</b>	180 Ringing
	Ringing tone		
	ANM ←	<b>←</b>	200 OK INVITE
		<b>→</b>	ACK
		onversation	
	REL ←	<b>←</b>	BYE
	RLC →	<b>→</b>	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)
		"network beyond interworking point"	
NOTE: "XX" is the Ca	use Value as defined in ITI	J-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/ Sendir	ISUP-SIP /Basic call/ Sending of the Release message (REL)/		
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	Final response without Reas	son header receive	ed, ma	oping in REL
SIP Parameter	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 is <b>not</b> included defined as SIP_Failure_VA:  sends a REL message with the Cause value set to CV_ISUP.			
values:				
ISUP Parameter values:	REL; cause value: CV_ISU	Р		
Comments:	ISUP/BICC	SUT		SIP
	IAM →		<b>→</b>	INVITE
	REL ←		<b>←</b>	SIP_Failure_VA
	RLC →		<b>→</b>	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:
			E	ES 283 027 [1], clause 7.2.3.1.7
TSS reference:	ISUP-SIP /Basic call/ S	ending of the Release	messa	ge (REL)/
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	Final response with Re			<u> </u>
				ut an INVITE message. On receipt of
	a Failure message (4xx			
		n Q.850 [5] Cause Valı	ue 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	REL; cause value: CV_ISUP			
values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>	→	INVITE
	REL(#xx)	<del>(</del>	<b>←</b>	SIP_Failure_VA
	RLC	<b>→</b>	<b>→</b>	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 contain	ns a Reason header in e	early dialogue received
SIP Parameter	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.  CV_SIP (PIXIT)		
values:			
ISUP Parameter	CV_ ISUP (PIXIT)		
values:			
Comments:	ISUP/BICC	SUT	SIP
	IAM	<b>→</b>	→ INVITE
			← SIP MESSAGE_VA
	REL	<b>←</b>	<ul><li>SIP_Failure_VA</li></ul>
	RLC	<b>→</b>	→ 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: I	REC 3261 [6]		ISUP reference:
11 000000		(1 O 0201 [0]		ES 283 027 [1], clause 7.2.3.1.7
TCC reference:	ICLID CID /Dania call/ Ca	anding of the Delega		
TSS reference:	ISUP-SIP /Basic call/ Se	ending of the Release	messa	ge (REL)/
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	Final response without I	Reason header receiv	ed, ma	oping in REL
SIP Parameter values:	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.			
ISUP Parameter	REL; cause value: 127			
values:	,			
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	REL(#127)	<b>←</b>	<b>←</b>	SIP_Response_VA
	RLC	<b>→</b>	<b>→</b>	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:	
11 303001	on reference. At 6 0201 [0]	ES 28	3 027 [1], clause 7.2.3.2.12.1	
TSS reference:	ISUP-SIP/Basic call/Autonomous release/			
SIP selection	PICS 3/2			
criteria:				
ISUP selection				
criteria:				
Test purpose:	Overlap supported, REL is sent when 404/484 received and Ti/w3 is expired			
	Ensure that the SUT on receipt of a 484 Address Incomplete or 404 Not Found response			
	for the current INVITE (i.e. there are no			
	the SUT is configured to propagate over			
	line con is configured to propagate over	map signaming into	alo on howork, the oot.	
	Shall not send a REL message implementation.	mediately and sha	II instead start timer TOIW3. The	
	REL message shall only be sent if			
	The REL message contains the Ca			
SIP Parameter				
values:				
ISUP Parameter				
values:				
Comments:		SUT	SIP	
	IAM →	<b>→</b>	INVITE	
	CASE A			
	CASE A	_	404 Address Insernate	
		<b>←</b>	484 Address Incomplete ACK	
	Stort tir	-	ACK	
	Start til	ner T <sub>I/W3</sub>		
	Time	T		
		ut T <sub>I/W3</sub>		
	REL #28 ←			
	RLC →			
	2.22			
	CASE B	-	404 Net Ferred	
		<del>(</del>	404 Not Found	
	01	<b>→</b>	ACK	
	Start tir	ner T <sub>I/W3</sub>		
	Timed	ut T <sub>I/W3</sub>		
	REL #28 ←	1/ ۷ ۷ 3		
	RLC →			
	INCO			

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 40	04/484 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 <b>←</b>	<ul> <li>484 Address Incomplete</li> </ul>	
	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	PICS 4/5 AND PICS 4/11	
criteria:		
ISUP selection	PICS 4/2	
criteria:		
Test purpose:	Preconditions supported, call setup released wh	nen COT(failed) received
	Ensure that the SUT on receipt of a COT "failed	" and preconditions used, the SUT:
	<ul> <li>sends a CANCEL to the SIP network.</li> </ul>	
SIP Parameter		
values:		
ISUP Parameter	IAM: Nature of connection indicators "continui	ty check required on this circuit"
values:		·
Comments:	ISUP/BICC SUT	SIP
	IAM →	→ INVITE
	COT(failed) →	→ CANCEL
		← 200 OK CANCEL
		← 487 Request terminated
		→ ACK
		2 /1011

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 v	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	sends a CANCEL or BYE to the SIP netw	OTK.	
values:			
ISUP Parameter values:	IAM: Nature of connection indicators "contin	uity check required on this circuit"	
Comments:	ISUP/BICC SUT	SIP	
	IAM →	→ INVITE	
	T8 expirees		
		→ CANCEL	
		← 200 OK CANCEL	
		<ul> <li>487 Request terminated</li> </ul>	
		→ ACK	

TP309005	SIP reference: R	FC 3261 [6]	E	ISUP reference: S 283 027 [1], clause 7.2.3.2.16
TSS reference:	ISUP-SIP/Basic call/Auto	nomous 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	
SIP Parameter	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			
values:				
ISUP Parameter values:	IAM: Nature of connection indicators "continuity check required on this circuit"			
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	REL	<b>←</b>	<b>←</b>	580 Precondition Failure
	RLC	<b>→</b>	<b>→</b>	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 Ensure that the SUT when the resource reserused, the SUT responds to an UPDATE:  send a REL with cause value # 47		
SIP Parameter values:	Solid d NEZ Will Eddoo Falde II		
ISUP Parameter values:	IAM: Nature of connection indicators "contin	uity check required on this circuit"	
Comments:	ISUP/BICC SUT	SIP	
	IAM →	→ INVITE	
		<ul> <li>183 Session Progress</li> </ul>	
		→ PRACK	
		← 200 OK PRACK	
	REL ←	<ul> <li>580 Precondition Failure</li> </ul>	
	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 referen	ce: 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:	ico: on /Badio da	iii reddipt di r	ooc on our m	occago (NOC)
ISUP selection criteria:				
Test purpose:	RSC received while	e an INVITE w	as not sent	·
OID D	receipt of a RSC m	essage: quired on the S		t before an INVITE has been sent on than to terminate local procedures if any
SIP Parameter values:				
ISUP Parameter values:				
Comments:	ISUP/BICC		SUT	SIP
	IAM	<b>→</b>		
	RSC RLC	<b>→</b>		

TP310002	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 no response for an INVIT Ensure that the SUT after receiving the IAM wish sending a INVITE message on receipt RSC missions message has been received:  The SUT shall hold the RSC message undirection to the SUT shall send a CANCEL request. A Reason header field containing the (ITU Value # 31 is added to the SIP message)	with the complete called party number, nessage <b>before</b> a <b>SIP MESSAGE_VA</b> ntil a SIP response has been received.	
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP/BICC SUT  IAM →  RSC →  RLC ←	SIP → INVITE  ← SIP MESSAGE_VA → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK	

Table 19

Values for test purpose TP310002			
VA	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		

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:	<ul> <li>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:         <ul> <li>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.</li> <li>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.</li> </ul> </li> </ul>		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP/BICC SI	JT	SIP
	IAM →	=	→ INVITE
		•	► 100 Trying
	RSC →		➤ CANCEL
	RLC ←		200 OK INVITE
			→ ACK
			200 OK CANCEL
			<b>BYE</b>
			€ 200 OK BYE

TP310004	SIP reference: RFC 3261 [6]		ISUP reference:	
TSS reference:	ES 283 027 [1], clause 7.2.3.2.15  ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC)			
SIP selection	130F-31F/Basic call/ Receipt of Reset	circuit iriessa	age (NSC)	
criteria:				
ISUP selection				
criteria:				
Test purpose:	RSC received after the ACK for a 200	OK INVITE V	vas sent. A BYE is sent	
SIP Parameter values: ISUP Parameter	<ul> <li>The SUT shall send a BYE reques</li> <li>A Reason header field containing</li> </ul>	mplete calle er a 200 OK .t. the (ITU-T R	d party number, sending a BYE response message has been received:	
values:				
Comments:	ISUP/BICC SU	T	SIP	
	IAM →	<b>→</b>	INVITE	
	ACM ←	<b>←</b>	180 Ringing	
	ANM <b>←</b>	<b>←</b>	200 OK INVITE	
		<b>→</b>	ACK	
	RSC →	<b>→</b>	BYE	
	RLC <b>←</b>	+	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:	<ul> <li>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:</li> <li>The SUT shall send a CANCEL or BYE request.</li> <li>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.</li> </ul>		
SIP Parameter			
values:			
ISUP Parameter values:			
Comments:	ISUP/BICC SUT	SIP	
Comments.	IAM →	→ INVITE  SIP MESSAGE VA	
	RSC →	_	
	RLC ←		
		<ul> <li>→ CANCEL</li> <li>← 200 OK CANCEL</li> <li>← 487 Request terminated</li> <li>→ ACK</li> </ul>	

#### Table 20

	Values for test purpose; TP310005			
VA	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 receipt of GRS message:	IAM but before an INVITE has been sent on	
	<ul> <li>no action is required on the SIP side other than to terminate local procedures if any are in progress.</li> </ul>		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP/BICC SUT	SIP	
	IAM →		
	GRS →		
	GRA <b>←</b>		

TP311002	SIP reference: R	FC 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:	sending a INVITE messaresponse message has but the SUT shall but the SUT shall such that the SUT shall	r receiving the IAM wage on receipt GRS moeen received:  nold the GRS messaged a CANCEL requer field containing the	vith the complete called party number, nessage before SIP MESSAGE_VA
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP/BICC	SUT	SIP
	IAM	<b>→</b>	→ INVITE
	GRS	<b>→</b>	
	GRA	<b>←</b>	← SIP MESSAGE_VA
			→ CANCEL
			← 200 OK CANCEL
			<ul> <li>487 Request terminated</li> </ul>
			→ ACK

Table 21

	Values for test purpose TP311002			
VA	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: RF	C 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			<u> </u>	
criteria:				
ISUP selection				
criteria:				
Test purpose:	GRS received. While CAN	,		
			with the complete called party number	
		e on receipt GRS m	nessage <b>before</b> a 200 OK response	message
	has been received:			
	Th - OUT 11 h - 1 d 4h	- 000	-411 b b b b b b-	
	<ul> <li>The SUT shall hold the CANCEL is sent.</li> </ul>	e GRS message un	ntil a response has been received. A	
		siving 200 OK INIVIT	FE massages the SLIT shall send on	ACK for
	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	001111			
values:				
ISUP Parameter				
values:				
Comments:	ISUP/BICC	SUT	SIP	
	IAM =	•	→ INVITE	
			<ul><li>100 Trying</li></ul>	
	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:	The SUT shall send a BYE requ	he IAM with the complete callet fter a 200 OK est.  g the (ITU-T R	ne complete called party number, ed party number, sending a BYE response message has been received:	
SIP Parameter values:				
ISUP Parameter				
values:				
Comments:	ISUP/BICC S	UT	SIP	
	IAM →	•· →	<del></del>	
	ACM ←	÷		
	ANM ←	<u>-</u>	5 5	
		<b>→</b>		
	GRS →	<b>→</b>	BYE	
	GRA <b>←</b>	<del>-</del>	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:	<ul> <li>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:     </li> <li>The SUT shall send a CANCEL request.</li> <li>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.</li> </ul>				
SIP Parameter	value // or le daded to the en meedage	to so come sy the children of the children.			
values: ISUP Parameter					
values:					
Comments:	ISUP/BICC SUT IAM →	SIP → INVITE ← SIP MESSAGE_VA			
	GRS → ←	_			
		<ul> <li>→ CANCEL</li> <li>← 200 OK CANCEL</li> <li>← 487 Request terminated</li> <li>→ ACK</li> </ul>			

Table 22

	Values for test purpose 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:	
TSS reference:	IOUD OID/Daria and Da			S 283 027 [1], clause 7.2.3.2.15	
	ISUP-SIP/Basic call/ Receipt of Circuit group reset message (GRS)				
SIP selection					
criteria:					
ISUP selection					
criteria:	000 (	010 : 1 0 1 -	חער נ		
Test purpose:	GRS for more than one				
				M's sending an INVITE message for	
	bigger than "1":	receipt of a GRS mes	ssage v	vere the Range Parameter value is	
	bigger than 7.				
	• the SLIT shall sone	I a BYE requests for ea	och call	association	
SIP Parameter	une 301 Shall Seno	i a b i c requests for ea	icii caii	association.	
values:					
ISUP Parameter					
values:					
Comments:	ISUP/BICC	SUT		SIP	
Comments.		→	<b>→</b>		
	` '	<b>+</b>	÷	180 Ringing	
		<del>(</del>	<del>-</del>	200 OK INVITE	
	AINIVI	~	_	ACK	
			7	ACK	
	IAM(2)	<b>→</b>	<b>→</b>	INVITE(2)	
	ACM	<del>(</del>	<b>←</b>	180 Ringing	
	ANM	<del>(</del>	<b>←</b>		
			<b>→</b>	ACK	
				-	
	GRS(1)	<b>→</b>	<b>→</b>	BYE(1)	
	GRA	<b>←</b>	<b>←</b>	200 OK BYE	
			<b>→</b>	BYE(2)	
			<b>←</b>	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	e: RFC 3261 [6]		ISUP reference:
TSS reference:	ISUP-SIP/Basic call/ hardware failure orie		group b	ES 283 027 [1], clause 7.2.3.2.15 clocking message (CGB) with the indication
SIP selection criteria:	Inardware failure one	inted		
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 IAM CBG CGBA	→ → ←	SUT	SIP

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 Messa failure oriented"	· .			
Comments:	ISUP/BICC SUT IAM → CGB → CGBA ←	SIP → INVITE  ← 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 orien	ted		
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	<ul> <li>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:     </li> <li>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.</li> <li>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.</li> </ul>			
SIP Parameter values:				
ISUP Parameter	CGB/CGBA: Circuit Group Supervision Message Type Indicator coded as "hardware			
values:	failure oriented"			
Comments:	ISUP/BICC	SUT	SIP	
	IAM	<b>→</b>	→ INVITE	
			← 100 Trying	
	CGB	<b>→</b>	→ CANCEL	
	CGBA	<b>←</b>	← 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 IN				
	Ensure that the SUT after receiving the IAM				
	sending a INVITE message with the complete				
	message on receipt CGB message Circuit Gi				
	coded as "hardware failure oriented" after a	200 OK response message has been			
	received:				
	The CLIT shall send a DVF request				
	The SUT shall send a BYE request.  A Person header field containing the (ITLL T Personmendation O 950 (51) Cause.				
	<ul> <li>A Reason header field containing the (ITU-T Recommendation Q.850 [5]) Cause</li> <li>Value # 31 is added to the SIP message to be sent by the SIP side of the O-MGCF.</li> </ul>				
SIP Parameter	Value # 31 is added to the 311 message	to be sent by the Sir side of the O-MOCI .			
values:					
ISUP Parameter	CGB/CGBA: Circuit Group Supervision Message Type Indicator coded as "hardware				
values:	failure oriented"	age Type maicater occor as maranare			
Comments:	ISUP/BICC SUT	SIP			
	IAM →	→ INVITE			
	ACM ←	← 180 Ringing			
	ANM ←	€ 200 OK INVITE			
		→ ACK			
	CGB →	→ BYE			
	CGBA <b>←</b>	€ 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 hardware failure oriented	UP-SIP/Basic call/ Receipt of Circuit group blocking message (CGB) with the indication ardware failure oriented			
SIP selection criteria:					
ISUP selection criteria:					
SIP Parameter	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.				
ISUP Parameter values:	CGB/CGBA: Circuit Group Supervision Messa	age Type Indicator coded as "hardware			
Comments:	ISUP/BICC SUT IAM →  CGB →  CGBA ←	SIP → INVITE ← SIP MESSAGE_VA  → 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	e: RFC 3261 [6]	ES	ISUP reference: 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	nardware failure offer	iteu		
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 G failure oriented"	Froup Supervision Messa	age Type	Indicator coded as "hardware
Comments:	ISUP/BICC	SUT		SIP
Comments.	IAM(1)	→	_	INVITE(1)
	ACM	<b>É</b>	É	* *
	ANM	<b>←</b>	<b>+</b>	8 8
	AINIVI	•	_	ACK
	IAM(2)	<b>→</b>	<b>→</b>	INVITE(2)
	ACM	<b>←</b>	<b>←</b>	180 Ringing
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
	CGB(1)	<b>→</b>	<b>→</b>	BYE(1)
	CGBA	<del>(</del>	<b>←</b>	200 OK BYE
			<b>→</b>	BYE(2)
			+	200 OK BYE
NOTE: BYE(1) a	nd BYTE(2) possible re	ceived in reverse order.		

# 6.3 Interworking of supplemetary services

## 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 F	rom header not in the E.164 Format, no	
	Privacy header. Send Calling party number		
	Ensure that the SUT in the Idle state, on recei		
	<ul> <li>the SIP P-Asserted-Identity containing NDC+ SN has not been received;</li> </ul>	g a URI with an identity in the format "+" CC+	
	<ul> <li>the SIP From header field containing a NDC+ SN has not been received;</li> </ul>	a URI with an identity in the format "+" CC+	
	a Privacy header field has not been r	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	Nature of address findicator = 0000000		
values:			
ISUP Parameter			
values:			
Comments:	SIP SU	JT ISUP	
	INVITE ->	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
	Ringin	g tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conver	rsation	
	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_02	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/	L3 203 027 [1], clause 7.2.3.1.2.0
SIP selection	311 -1301 /33/3E1/	
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	
SIP Parameter		
values: ISUP Parameter		
values:		
Comments:	SIP SU	JT ISUP
	INVITE →	→ IAM
	180 Ringing ←	← ACM
	Ringin	g tone
	200 OK INVITE ←	← ANM
	ACK →	
	Conve	
	BYE →	→ REL
	200 OK BYE ←	<b>←</b> RLC

TP501003	SIP reference: RFC 3261 [6]	ı	SUP reference:
		ES 283 0	27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection			
criteria:			
ISUP selection			
criteria:	B.A	. , ,	·
Test purpose:	<ul> <li>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".</li> <li>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</li> </ul>		
SIP Parameter			
values:			
ISUP Parameter values:			
Comments:		JT	ISUP
	INVITE ->	<b>→</b>	IAM
	180 Ringing ←	+	ACM
	_	g tone	
	200 OK INVITE ←	<b>←</b>	ANM
	ACK →		
		rsation	
	BYE →	<b>→</b>	REL
	200 OK BYE ←	+	RLC

TP501004	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/	JI	F 47
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 user. 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 "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		
SIP Parameter			
values: ISUP Parameter			
values:			
Comments:	SIP SU	JT	ISUP
	INVITE →	<b>→</b>	IAM
	180 Ringing ←	<b>←</b>	ACM
	Ringin		
	200 OK INVITE ←	+	ANM
	ACK →		
	Conve		DEL
	BYE  200 OK BYE  ←	<b>→</b>	REL
	200 OK BYE ←	<u>+</u>	RLC

TP501005	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection			
criteria:			
ISUP selection			
criteria:	B.4		
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		
SIP Parameter			
values:			
ISUP Parameter values:			
Comments:	SIP SU	JT	ISUP
	INVITE ->	<b>→</b>	IAM
	180 Ringing ←	<b>←</b>	ACM
	Ringin		
	200 OK INVITE ←	+	ANM
	ACK →		
	Conve		DEL
	BYE →	<b>→</b>	REL
	200 OK BYE ←	<u> </u>	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 part  Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator Nature of address indicator = 0000000  with the Generic number parameter coded:  Address signals = derived from the From Screening indicator = user provided, not v Number Incomplete Indicator = complete Numbering plan indicator = ISDN numberi Address Presentation Restricted Indicator NoAS: NoA_VALUE	e = PIXIT  neader erified  ng plan	
SIP Parameter			
values: ISUP Parameter			
values:			
Comments:	SIP SI	JT ISUP	
Johnnents.	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
		g tone	
	200 OK INVITE ←	← ANM	
	ACK →		
		rsation	
	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_02	a different country	"International number"	CC+NDC+SN

TP501007	SIP reference: RFC 3261 [6]	I I	SUP reference:
		ES 283 02	27 [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 "+"		
	<ul> <li>CC+ NDC+ SN has not been received;</li> <li>the SIP From header field containin NDC+ SN has been received;</li> <li>a Privacy header field was received "none".</li> </ul>	<b>ved</b> ; g a URI with an i	dentity in the format "+" CC+
	sends an IAM message with the Calling par	ty number para	meter 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 <b>Generic number parameter</b> 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	UT	ISUP
	INVITE →	·. →	IAM
	180 Ringing ←	<del>-</del>	ACM
		ng tone	
	200 OK INVITE	<b>←</b>	ANM
	ACK →		
		ersation	
	BYE →	<b>→</b>	REL
	200 OK BYE ←	<b>←</b>	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_02	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/	L3 203 027 [1], Clause 7.2.3.1.2.0	
SIP selection	OII TOOT TOOT OLIT		
criteria:			
ISUP selection	PICS 6/3		
criteria:			
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 "+"		
	NDC+ SN has been received;	ved; g a URI with an identity in the format "+" CC+ and the priv-value component is set to	
	sends an IAM message with the Calling part	y number parameter coded:	
	Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplete Numbering plan indicator = 000 Address Presentation Restricted Indicator Nature of address indicator = 0000000 with the <b>Generic number parameter</b> coded:  Address signals = number provided by the Screening indicator = user provided, not v Number Incomplete Indicator = complete Numbering plan indicator = ISDN numberi Address Presentation Restricted Indicator NoAS: NoA_VALUE	e user verified	
SIP Parameter values:			
ISUP Parameter			
values:			
Comments:	SIP SU	UT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	_	ng tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
		rsation	
	BYE -	→ REL	
	200 OK BYE ←	<b>←</b> 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_02	a different country	"International number"	CC+NDC+SN

TP501009	SIP reference: RFC 3261 [6]		SUP reference: 27 [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:	<ul> <li>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:</li> <li>the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received;</li> <li>the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received;</li> <li>a Privacy header field was received and the priv-value component is set to "user".</li> </ul>		
	sends an IAM message with the Calling par Address signals = absent Screening indicator = network provided Number Incomplete Indicator = incomplet Numbering plan indicator = 000 Address Presentation Restricted Indicato Nature of address indicator = 0000000 with the Generic number parameter coded: Address signals = number provided by th Screening indicator = user provided, not y Number Incomplete Indicator = complete Numbering plan indicator = ISDN number Address Presentation Restricted Indicator NoAS: NoA_VALUE	e r = PIXIT e user verified ing plan	
SIP Parameter values:			
ISUP Parameter			
values:			
Comments:		UT	ISUP
	INVITE →	<b>→</b>	IAM
	180 Ringing ←	<b>←</b>	ACM
		ng tone	A N I N A
	200 OK INVITE	<b>←</b>	ANM
	ACK →		
		ersation	DEL
	BYE -	<b>→</b>	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_02	a different country	"International number"	CC+NDC+SN

TP501010	SIP reference: RFC 3261 [6]	ISUP reference:	
117301010	Sir Telefelice. NFC 3201 [0]	ES 283 027 [1], clause 7.2.3.1.2.6	
TSS reference:	SIP-ISUP/SS/CLI/	20 200 021 [1], 010000 11210111210	
SIP selection	011 1001 700/021/		
criteria:			
ISUP selection	PICS 6/3		
criteria:			
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 S	UT ISUP	
Comments.	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
		ng tone	
	200 OK INVITE ←	← ANM	
	ACK →	- / 11 1111	
		ersation	
	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_02	a different country	"International number"	CC+NDC+SN

TP501011	SIP reference: RFC 3261 [6]		SUP reference:	
		ES 283 0	27 [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 rece	ipt 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;			
	<ul> <li>the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received;</li> </ul>			
	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 S	UT	ISUP	
	INVITE →	<b>→</b>	IAM	
	180 Ringing ←	<b>←</b>	ACM	
	5 5	ng tone		
	200 OK INVITE ←	<b>+</b>	ANM	
	ACK →			
		rsation		
	BYE →	<b>→</b>	REL	
	200 OK BYE ←	<b>←</b>	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_02	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 SU	IT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
	Ringin		
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conver	rsation	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> 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_02	a different country	"International number"	CC+NDC+SN

TP501013	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		h 4/
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 SU	JT	ISUP
	INVITE →	<b>→</b>	IAM
	180 Ringing ←	<b>←</b>	ACM
	Ringin	g tone	
	200 OK INVITE ←	<b>←</b>	ANM
	ACK →		
	Conver	sation	
	BYE →	<b>→</b>	REL
	200 OK BYE ←	<b>←</b>	RLC

TP501014	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection			
criteria:			
ISUP selection			
criteria:			
Test purpose:	<ul> <li>P-Asserted-Identity in E.164 format and From header not in the E.164 Format, Privacy value user received. Send Calling party number</li> <li>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</li> <li>the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received;</li> <li>the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received;</li> <li>a Privacy header field was received and the priv-value component is set to "user".</li> <li>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</li> </ul>		
	Address Presentation Restricted Indicator	<ul><li>Presentation</li></ul>	restricted
SIP Parameter values:			
ISUP Parameter			
values:			
Comments:	SIP SU		ISUP
	INVITE →	<b>→</b>	IAM
	180 Ringing ←	<b>←</b>	ACM
	Ringin		
	200 OK INVITE ←	<b>←</b>	ANM
	ACK →		
	Conver	sation	
	BYE →	<b>→</b>	REL
	200 OK BYE ←	+	RLC

TP501015	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection	0.1.1001700702.11		
criteria:			
ISUP selection			
criteria:			
Test purpose:	<ul> <li>P-Asserted-Identity in E.164 format and From header not in the E.164 Format, Privacy value id received. Send Calling party number</li> <li>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</li> <li>the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received;</li> <li>the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received;</li> <li>a Privacy header field was received and the priv-value component is set to "id".</li> <li>sends an IAM message with the Calling party number parameter coded:</li> <li>Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = complete</li> <li>Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted</li> </ul>		
SIP Parameter			
values:			
ISUP Parameter			
values:			
Comments:	SIP SU		ISUP
	INVITE →	<b>→</b>	IAM
	180 Ringing ←	<b>←</b>	ACM
	Ringing		
	200 OK INVITE ←	+	ANM
	ACK →		
	Conver		
	BYE →	<b>→</b>	REL
	200 OK BYE ←	+	RLC

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		
SIP Parameter			
values: ISUP Parameter			
values:			
Comments:	SIP SU	UT ISUP	
	INVITE ->	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
		ng tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conve	ersation	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> 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_02	a different country	"International number"	CC+NDC+SN

TP501017	SIP reference: RFC 3261 [6]	ISUP reference:	
1501017	Sir leierence. Krc 3201 [0]	ES 283 027 [1], clause 7.2.3.1.2.6	
TSS reference:	SIP-ISUP/SS/CLI/	20 200 027 [1], olddod 7.2.0.1.2.0	
SIP selection	CII 1001 700/02I/		
criteria:			
ISUP selection	PICS 6/3		
criteria:			
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 part Address signals = number derived from S Screening indicator = network provided Number Incomplete Indicator = complete Numbering plan indicator = ISDN number Address Presentation Restricted Indicator NoAS: NoA_VALUE  with the Generic number parameter coded: Address signals = number derived from the Screening indicator = user provided, not we Number Incomplete Indicator = complete Numbering plan indicator = ISDN number Address Presentation Restricted Indicator NoAS: NoA_VALUE	ing plan = Presentation allowed  e From header erified	
SIP Parameter			
values: ISUP Parameter			
values:			
Comments:	SIP SI	JT ISUP	
Johnner 13.	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
		ig tone	
	200 OK INVITE ←	€ ANM	
	ACK →		
		rsation	
	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_02	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 SI	UT ISUP		
	INVITE →	→ IAM		
	180 Ringing ←	<b>←</b> ACM		
		ng tone		
	200 OK INVITE ←	<b>←</b> ANM		
	ACK →			
		ersation		
	BYE →	→ REL		
	200 OK BYE ←	<b>←</b> 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_02	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 NoAS: NoA_VALUE	= Presentation restricted	
SIP Parameter values:			
ISUP Parameter			
values:	SIP SI	UT ISUP	
Comments:	INVITE →	→ IAM	
	180 Ringing	→ IAIM ← ACM	
		- /	
	200 OK INVITE ←	ng tone ← ANM	
	ACK →	VIAINI	
	-	ersation	
	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_02	a different country	"International number"	CC+NDC+SN

TP501020	SIP reference: RFC 3261 [6]		JP 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 Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan		
SIP Parameter values:			
ISUP Parameter			
values:			
Comments:	SIP SU	T	ISUP
	INVITE ->		IAM
	180 Ringing ←	<b>←</b>	ACM
	Ringin	g tone	
	200 OK INVITE ←		ANM
	ACK →		
	Conve	sation	
	BYE →	<b>→</b>	REL
	200 OK BYE ←	+	RLC

TP501021	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.2.3.1.2.6		
TSS reference:	SIP-ISUP/SS/CLI/	E3 203 U	21 [1], clause 1.2.3.1.2.6		
SIP selection	51P-15UP/55/CLI/				
criteria:					
ISUP selection	PICS 6/1				
criteria:	PICS 0/ I				
Test purpose:	P-Asserted-Identity not in E.164 format and Fi	rom hooder not	in the E 164 Format no		
rest purpose.	Privacy header. Send Calling party number ne				
	Ensure that the SUT in the Idle state, on recei				
	Lindie that the 501 in the late state, on recei	pt of a liveric	message where.		
	the SIP P-Asserted-Identity containing	ı a URI with an	identity in the format "+" CC+		
	NDC+ SN has not been received:	, a ora maran	radinary in the format		
	the SIP From header field containing a	a URI with an ic	dentity in the format "+" CC+		
	NDC+ SN has not been received:		201111, 111 110 101111011 7 007		
	a Privacy header field has not been re	eceived.			
	sends an IAM message with the Calling party	number para	meter coded:		
	Address signals = network provided (PIXIT	<u> </u>			
		Screening indicator = network provided			
	Nature of address indicator = NoA_VALUE				
	Number Incomplete Indicator = complete				
	Numbering plan indicator = ISDN numbering plan				
OID D	Address Presentation Restricted Indicator	= PIXII			
SIP Parameter					
values:					
ISUP Parameter values:					
Comments:	SIP SU	·	ISUP		
Comments:	J	'' →	IAM		
		<b>→</b>	ACM		
	100 1 11191119	=	ACIVI		
	Ringing		A N I N 4		
	200 OK INVITE	+	ANM		
	ACK →				
	Conver				
	BYE -	<b>→</b>	REL		
	200 OK BYE ←	+	RLC		

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 SU		
	INVITE →	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
	Ringin	g tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conve	rsation	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	

TP501023	SIP reference: RFC 3261 [6]	ISUP refere ES 283 027 [1], claus	
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection			
criteria:			
ISUP selection	PICS 6/1 AND PICS 6/12		
criteria:	D. Assertad I dentity not in E. 404 famous to as D		
Test purpose:	P-Asserted-Identity not in E.164 format, no P network provided Address not available Ensure that the SUT in the Idle state, on rece  the SIP P-Asserted-Identity containin NDC+ SN has not been received; a Privacy header field has not been sends an IAM message with the Calling part Address signals = not present Screening indicator = network provided Number Incomplete Indicator = incomplet Address Presentation Restricted Indicator NoAS: NoA_VALUE	pt of a INVITE message when a URI with an identity in the eceived.  y number parameter code	nere: ne format "+" CC+
SIP Parameter			
values:			
values:			
Comments:	SIP S	JT ISUP	
	INVITE ->	→ IAM	
	180 Ringing ←	← ACM	
	Ringir	g tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
		rsation	
	BYE -	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

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;				
	NDC+ SN has been received;  a Privacy header field has not been	a URI with an identity in the format "+" CC+ received.			
	sends an IAM message with the Calling part	y 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 S	JT ISUP			
	INVITE ->	→ IAM			
	180 Ringing ←	← ACM			
	200 OK INVITE ← ACK →	<b>←</b> ANM			
		rsation			
	BYE → 200 OK BYE ←	→ REL ← RLC			

Table 37: Values for test purposes TP501022, 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_02	a different country	"International number"	CC+NDC+SN

## 6.3.1.2 Call Hold (HOLD)

TP502001	SIP reference: RFC 326	1 [6]		UP reference: use 7.4.10/ [14]
TSS reference:	SIP-ISUP/SS/HOLD/		O.C.	400 7.4.10/ [14]
SIP selection	PICS 8/4			
criteria:				
ISUP selection	PICS 5/22			
criteria:				
Test purpose:	Each party can hold and retrieve	the remote p	arty in the confir	med 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 t			
SIP Parameter	SDP: a=sendonly (put on hold)			
values:	a=sendrecv or omitted (re		)	
ISUP Parameter	o= <version incremente<="" th=""><th></th><th>indicator DDOOL</th><th></th></version>		indicator DDOOL	
values:	CPG: Generic notification: remote Generic notification: remote CPG: Generi			
Comments:	SIP		CF	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>(-</b>	<b>←</b>	ACM
		←	<b>←</b>	ANM
		<b>→</b> ←	<b>→</b>	CPG(hold)
		<b>→</b> ←	<b>→</b>	CPG(retrieve)
		<del>←</del> →	<b>←</b>	CPG(hold)
		<del>←</del> →	+	CPG(retrieve)

TP502002	SIP reference: RFC 33	261 [6]		SUP reference: ause 7.4.10/ [14]
TSS reference:	SIP-ISUP/SS/HOLD/		0.0	2400 714.10/ [14]
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	SDP: a=sendonly (put on hole		, , ,	
values:	a=sendrecv or omitted (retrieve the call) o= <version incremented=""></version>			
ISUP Parameter values:	CPG: Generic notification: rem Generic notification: rem			RESS (put on hold) OGRESS (retrieve the call)
Comments:	SIP	MC	SCF	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM
	UPDATE(sendonly) 200 OK UPDATE(recvonly)	<b>→</b> ←	<b>→</b>	CPG(hold)
	UPDATE(sendrecv) 200 OK UPDATE(sendrecv)	<b>→</b> ←	<b>→</b>	CPG(retrieve)

TP502003	SIP reference: RFC 3261 [6]	ISUP reference: clause 7.4.10/[14]
TSS reference:	SIP-ISUP/SS/HOLD/	
SIP selection	PICS 8/2	
criteria:		
ISUP selection	PICS 5/22	
criteria:		
Test purpose:	The calling party can hold and retrieve the renall information to process the call	note party after the calling party has provided
	Ensure that a party can put the other party on	
	the information necessary for processing the o	call. Ensure that the party can retrieve the
	call previously put on hold.	
	The colling party should be able to put the oth	or party on hold
	The calling party should be able to put the oth The calling party should be able to retrieve the	
SIP Parameter	SDP: a=sendonly (put on hold)	o diloi party
values:	a=sendrecv or omitted (retrieve the cal	1)
	o= <version> incremented</version>	,
ISUP Parameter	ACM: called party status: no indication	
values:	CPG: Generic notification: remote hold Event	
		ent indicator PROGRESS (retrieve the call)
Comments:	1	GCF ISUP
	INVITE ->	→ IAM
	LIBBATE( L L)	
	UPDATE(sendonly) → 200 OK UPDATE(recyonly) ←	
	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	PICS 8/4		
criteria:			
ISUP selection	PICS 5/22		
criteria:			
Test purpose:	A party can hold and retrieve the remote party 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 oth The calling party should be able to retrieve th	1 7	
SIP Parameter	SDP: a=sendonly (put on hold)		
values:	a=sendrecv or omitted (retrieve the ca o= <version incremented=""></version>	all)	
ISUP Parameter	CPG: Generic notification: remote hold Event	indicator PROGRESS (put on hold)	
values:		ent indicator PROGRESS (retrieve the call)	
Comments:		GCF ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	200 OK INVITE ←	<b>←</b> 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	SDP: a=sendonly (put on hold)		
values:	a=sendrecv or omitted (retrieve t	he call)	
	o= <version incremented=""></version>		
ISUP Parameter	CPG: Generic notification: remote hold		
values:	Generic notification: remote retriev	al event indicator P	PROGRESS (retrieve the call)
Comments:	SIP	MGCF	ISUP
	INVITE →	-	IAM
	180 Ringing ←	•	- ACM
	200 OK INVITE ←	•	- ANM
	UPDATE(sendonly) ← 200 OK INVITE(recvonly) →	•	- CPG(hold)
	UPDATE(sendrecv) ← 200 OK UPDATE(recvonly) →	•	- CPG(retrieve)

TP502006	SIP reference: RFC 3	3261 [6]	IS	SUP reference: 7.4.10/[14]	
TSS reference:	SIP-ISUP/SS/HOLD/		•		
SIP selection criteria:	PICS 8/4				
ISUP selection	PICS 5/22				
criteria:					
Test purpose:	Both parties can hold and retrieve the remote party in the confirmed state. First 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 calling party should be able to retrieve the other party  The called party should be able to retrieve the other party				
SIP Parameter values:	SDP: a=sendonly or a=inactive (put on hold) a=sendrecv or a=recvonly or omitted (retrieve the call) o= <version incremented=""></version>				
ISUP Parameter values:	CPG: Generic notification: rer	mote hold Event		RESS (put on hold) OGRESS (retrieve the call)	
Comments:	SIP	M	GCF	ISUP	
	INVITE	<b>→</b>	<b>→</b>		
	180 Ringing 200 OK INVITE	<b>←</b> <b>←</b>	<b>+</b>		
	INVITE(sendonly) 200 OK INVITE(recvonly)	<b>→</b>	<b>→</b>	CPG(hold)	
	INVITE(inactive) 200 OK INVITE(inactive)	<b>←</b> →	<b>←</b>	CPG(hold)	
	INVITE(recvonly) 200 OK INVITE(sendonly)	<b>→</b> ←	<b>→</b>	CPG(retrieve)	
	INVITE(sendrecv) 200 OK INVITE(sendrecv)	<b>←</b> →	<b>←</b>	CPG(retrieve)	

TP502007	SIP reference: RFC 3261	l [6]	IS	SUP 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:	Both parties 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 retrieve the other party				
	The calling party should be able t		e other party		
SIP Parameter values:	SDP: a=sendonly or a=inactive (put on hold) a=sendrecv or a=recvonly or omitted (retrieve the call) o= <version incremented=""></version>				
ISUP Parameter values:	CPG: Generic notification: remote Generic notification: remote				
Comments:	SIP		SCF	ISUP	
	INVITE -	<b>→</b>	→		
	1	F	<b>←</b>		
	200 OK INVITE	<b>-</b>	+		
		<b>→</b> <del>-</del>	<b>→</b>	CPG(hold)	
		<b>←</b> <b>→</b>	<b>←</b>	CPG(hold)	
		<b>←</b> <b>→</b>	<b>←</b>	CPG(retrieve)	
		<b>→</b> +	<b>→</b>	CPG(retrieve)	

# 6.3.1.3 Terminal portability (TP)

Void.

# 6.3.1.4 Conference calling (CONF)

TP504001	SIP reference: RFC 3261 [6]	E	NGN reference: S 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 Conference established and Conference disconnected and SIP procedure			
	Ensure that the SUT does not stop the ter streams if a CPG message Generic notific supplementary service.			
SIP Parameter values:				
ISUP Parameter values:	CPG: Generic notification = Conference of CPG: Generic notification = CP			
Comments:	SIP	SUT	ISUP	
	INVITE →	<b>→</b>	IAM	
	180 Ringing ←	<b>←</b>	ACM	
	Ringing tone			
	200 OK INVITE ←	+	ANM	
	ACK →			
	Со	nversation	000/0 / / / / / / /	
	6-		CPG(Conference established)	
	Co	nversation <b>←</b>	CPG(Conference disconnected)	
	BYE →	<b>→</b>	REL	
	200 OK BYE <b>←</b>	+	RLC	

TP504002	SIP reference: RFC 3261 [6]	_	NGN reference:		
		E	S 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 <b>GEN_NOT_VALUE</b> was received due to the CONF supplementary service.				
	If the media stream is either in state "sendon line a=sendonly/sendrecv or omitted attribu				
SIP Parameter	SDP: a=sendonly/sentrecv or a line is omi	tted	-		
values:					
ISUP Parameter	CPG: Generic notification = Conference esta	ablished			
values:	CPG: Generic notification = Isolated CPG: Generic notification = Reattached				
	CPG: Generic notification = Conference disc	connected			
Comments:	SIP SU	Γ	ISUP		
	INVITE →	<b>→</b>	IAM		
	180 Ringing ←	<b>←</b>	ACM		
	Ringing tone				
	200 OK INVITE ← ACK →	<b>←</b>	ANM		
	-	ersation			
	33		CPG(Conference established)		
	INVITE(sendonly) 200 OK INVITE(recvonly)  ACK  ←	+	CPG(Isolated)		
	INVITE(sendrecv)  200 OK INVITE(sendrecv)  ACK	<b>←</b>	CPG(Reattached)		
	Conve	ersation <b>←</b>	CPG(Conference disconnected)		
	BYE → 200 OK BYE ←	<b>→</b>	REL RLC		

TP504003	SIP reference: RFC 3261 [6]	E	NGN reference: S 283 027 [1], clause 7.4.14			
TSS reference:	SIP-ISUP/SS/CONF/					
SIP selection	NOT PICS 5/10	NOT PICS 5/10				
criteria:						
ISUP selection						
criteria:	No many in a stip of the design of the design of					
Test purpose:	No mapping of isolated and reattached					
	Ensure that the SUT on receipt of a CPG mes	ssage due	to the CONF supplementary			
	service, the Generic notification indicator with	the value				
	No mapping, no disrupting the SIP proced	ure.				
SIP Parameter	No mapping					
values:						
ISUP Parameter values:	CPG: Generic notification = Conference esta CPG: Generic notification = isolated	iblished				
values:	CPG: Generic notification = isolated					
	CPG: Generic notification = Conference disc	onnected				
Comments:	SIP SUT		ISUP			
	INVITE →	<b>→</b>	IAM			
	180 Ringing ←	<b>←</b>	ACM			
	Ringing tone					
	200 OK INVITE ←	+	ANM			
	ACK →					
	Conve	rsation	ODO(0			
		<b>←</b>	CPG(Conference established)			
		<b>←</b>	CPG(Isolated)			
		. +	CPG(Reattached)			
	Conve	rsation <b>←</b>	CPG(Conference disconnected)			
	BYE →	<b>→</b>	REL			
	200 OK BYE ←	<u>+</u>	RLC			

TP504004	SIP reference: RFC 3	261 [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 notification	ations no char	nge the ses	sion 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	MG	CF.	ISUP	
	INVITE	→	→	IAM	
	180 Ringing	<del>-</del>	<b>+</b>	ACM	
	200 OK INVITE	÷	÷	ANM	
	ACK	<b>→</b>	•	AUVI	
			<b>←</b>	CPG(Conference established)	
			<b>←</b>	CPG(other party added)	
			<b>←</b>	CPG(other party isolated)	
			<b>←</b>	CPG(other party reattached)	
			<b>←</b>	CPG(other party split)	
			<b>←</b>	CPG(other party disconnected)	
			<b>←</b>	CPG(Conference floating)	
			<b>←</b>	CPG(Conference disconnected)	
	BYE	<b>←</b>	<b>←</b>	REL	
	200 OK BYE	<b>→</b>	→	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	l into "co	nference 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".</conference-state-type>			
SIP Parameter	NOTIFY 1: Event contains conference; Su	bscriptio	n-State contains active;	
values:	expires=xxxx			
	application/conference-info+xm	l:		
	<conference-info></conference-info>			
	entity=conference URI s	tate="full	" version="x"	
	<conference-state></conference-state>	oounts	if present	
	<pre><user-count>2true</user-count></pre>			
	<users></users>	> II piese	5111	
	<user entity="ISUPx" th="" u<=""><th>JRI state</th><th>="full"</th></user>	JRI state	="full"	
	<pre><endpoint entity="&lt;/pre"></endpoint></pre>			
	<status>conn</status>			
			d-out joining-method	
	<media <="" id="1" th=""><th>1</th><th></th></media>	1		
	<status>s</status>	endrecv<	:/status>	
ISUP Parameter				
values:				
Comments:	SIP MGCF		ISUP	
	INVITE -	<b>→</b>	IAM	
	180 Ringing ←	<b>←</b>	ACM	
	200 OK INVITE ←	+	ANM	
	ACK →			
	INVITE(SDP focus) →			
	200 OK INVITE			
	ACK →			
	NOTIFY 1 →	<b>→</b>	CPG(Conference established)	
	BYF <b>←</b>	+	REL	
	200 OK BYE →	<b>→</b>	RLC	

TP504006	SIP reference: RI	FC 3261 [6]		NGN reference: ES 283 027 [1], clause 7.5.6		
TSS reference:	SIP-ISUP/SS/CONF/		<u> </u>	20 203 027 [1], clause 7.3.0		
SIP selection	PICS 1/1					
criteria:	1100 1/1					
ISUP selection						
criteria:						
Test purpose:	Conference notification information is mapped into "other party added"					
	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".</endpoint-type>					
SIP Parameter	NOTIFY 1: see test case					
values:				n-State contains active		
		conference-info+xm	ıl:			
		ence-info>				
		y=conference URI s nference-state>	tate="tull	version="X"		
		:user-count>3 <th>r-count&gt; it</th> <th>nresent</th>	r-count> it	nresent		
	<use< th=""><th></th><th>i count&gt; ii</th><th>present</th></use<>		i count> ii	present		
		<i>user</i> entity=SIPx UI	RI state="	full"		
		<endpoint entity="&lt;/th"><th></th><th></th></endpoint>				
		<status>conr</status>				
				d-out joining-method		

TP504007	SIP reference: RFC 32	261 [6]	E	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".</endpoint-type>				
SIP Parameter values:	NOTIFY 1: see test case 504005  NOTIFY 2: Event contains <b>conference</b> ; Subscription-State contains <b>active</b> application/conference-info+xml:				
ISUP Parameter values:			sendrecv<		
Comments:	SIP INVITE 180 Ringing 200 OK INVITE ACK INVITE(SDP focus) 200 OK INVITE ACK	MGCF → ← ← → →	÷ + +	ISUP IAM ACM ANM	
	NOTIFY 1 200 OK NOTIFY	<b>→</b>	<b>→</b>	CPG(Conference established)	
	NOTIFY 2 200 OK NOTIFY	<b>→ ←</b>	<b>→</b>	CPG(isolated)	
	BYE 200 OK BYE	<b>←</b> →	<b>←</b> →	REL RLC	

TP504008	SIP reference: RFC 3261 [6	<b>6</b> ]	F	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	io monno	d into "oth	er party isolated"	
SIP Parameter values:	Conference notification information is mapped into "other party 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 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".  NOTIFY 1: See test case 504005</endpoint-type>				
ISUP Parameter	application/conference-info+xml:				
values:					
Comments:	SIP INVITE →	MGCF	÷ →	ISUP IAM	
	180 Ringing ← 200 OK INVITE ← ACK →		<del>+</del>	ACM ANM	
	INVITE(SDP focus)  200 OK INVITE  ACK  →				
	NOTIFY 1 → 200 OK NOTIFY ←		<b>→</b>	CPG(Conference established)	
	NOTIFY 2 200 OK NOTIFY ←		<b>→</b>	CPG(other party isolated)	
	BYE		<b>←</b> →	REL RLC	

	SIP reference: F	RFC 3261 [6]		NGN reference: ES 283 027 [1], clause 7.5.6				
TSS reference:	SIP-ISUP/SS/CONF/			23 263 027 [1], clause 7.3.0				
SIP selection	PICS 1/1							
criteria:								
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".</endpoint-type>							
SIP Parameter		ase 504005		0.1				
values:				n-State contains <b>active</b>				
		n/conference-info+xm	l <b>:</b>					
		erence-info>	toto-"full	"vorsion_"v"				
		tity=conference URI s <sup>.</sup> onference-state>	iaie= iuii	version= x				
		<user-count>3<th>-counts i</th><th>f nresent</th></user-count>	-counts i	f nresent				
	<11	Sers>	50u1121	Prodont				
		<i><user< i=""> entity=ISUPx \</user<></i>	JRI state	="full"				
		<endpoint entity="&lt;/th"><th></th><th></th></endpoint>						
		<status>on-h</status>						
		, ,		d-out joining-method				
		<media <="" id="1" th=""><th></th><th>latation.</th></media>		latation.				
	NOTIFY 3: Event con	<pre>status&gt;s</pre>		r/status> n-State contains <b>active</b>				
		n/conference-info+xm		1-State contains active				
		erence-info>						
		tity=conference URI s	tate="full	" version="x"				
		onference-state>						
		<user-count>3<th>-count&gt; i</th><th>f present</th></user-count>	-count> i	f present				
	<u< th=""><th>sers&gt;</th><th></th><th></th></u<>	sers>						
		<user entity="ISUPx" l<="" th=""><th></th><th></th></user>						
		<endpoint entity="&lt;/th"><th></th><th></th></endpoint>						
				<status>connected</status>				
	<pre><joining-method>dialed-out</joining-method></pre> / joining-method>							
	<media <status="" id="1">sendrecv</media>							
		<media <="" id="1" th=""><th>1</th><th></th></media>	1					
ISUP Parameter	_	<media <="" id="1" th=""><th>1</th><th></th></media>	1					
10011111111111111		<media <="" id="1" th=""><th>1</th><th></th></media>	1					
ISUP Parameter values: Comments:	SIP	<media <br="" id="1"><status>s</status></media>	endrecv<	:/status>				
values:	SIP INVITE	<media <="" id="1" th=""><th>endrecv&lt;</th><th>/status&gt;</th></media>	endrecv<	/status>				
values:	INVITE	<media <status="" id="1">so</media>	endrecv<	isup IAM				
values:	INVITE 180 Ringing	<media 1"="" <status="" id="1' &lt;status&gt;se  MGCF →&lt;/th&gt;&lt;th&gt;endrecv&lt;&lt;/th&gt;&lt;th&gt;isup&lt;br&gt;IAM&lt;br&gt;ACM&lt;/th&gt;&lt;/tr&gt;&lt;tr&gt;&lt;th&gt;values:&lt;/th&gt;&lt;th&gt;INVITE&lt;br&gt;180 Ringing&lt;br&gt;200 OK INVITE&lt;/th&gt;&lt;th&gt;&lt;media id=">si  MGCF →  ←</media>	endrecv<	isup IAM				
values:	INVITE 180 Ringing	<media <status="" id="1">si  MGCF</media>	endrecv<	ISUP IAM ACM				
values:	INVITE 180 Ringing 200 OK INVITE	<media <status="" id="1">si  MGCF</media>	endrecv<	ISUP IAM ACM				
values:	INVITE 180 Ringing 200 OK INVITE ACK	<media <status="" id="1">si  MGCF</media>	endrecv<	isup IAM ACM				
values:	INVITE 180 Ringing 200 OK INVITE ACK INVITE(SDP focus)	<media <status="" id="1">si  MGCF  →  ←  ←  →  →</media>	endrecv<	ISUP IAM ACM				
values:	INVITE 180 Ringing 200 OK INVITE ACK INVITE(SDP focus) 200 OK INVITE	<media <status="" id="1">si  MGCF</media>	endrecv<	isup IAM ACM				
values:	INVITE 180 Ringing 200 OK INVITE ACK INVITE(SDP focus) 200 OK INVITE ACK NOTIFY 1	<media <status="" id="1">si  MGCF</media>	endrecv<	isup IAM ACM				
values:	INVITE 180 Ringing 200 OK INVITE ACK INVITE(SDP focus) 200 OK INVITE ACK	<media <status="" id="1">si  MGCF </media>	endrecv<	ISUP IAM ACM ANM				
values:	INVITE 180 Ringing 200 OK INVITE ACK  INVITE(SDP focus) 200 OK INVITE ACK  NOTIFY 1 200 OK NOTIFY	<media <status="" id="1">si  MGCF</media>	endrecv<	ISUP IAM ACM ANM  CPG(Conference established)				
values:	INVITE 180 Ringing 200 OK INVITE ACK  INVITE(SDP focus) 200 OK INVITE ACK  NOTIFY 1 200 OK NOTIFY  NOTIFY 2	<media <status="" id="1">si  MGCF</media>	endrecv<	ISUP IAM ACM ANM				
values:	INVITE 180 Ringing 200 OK INVITE ACK  INVITE(SDP focus) 200 OK INVITE ACK  NOTIFY 1 200 OK NOTIFY	<media <status="" id="1">si  MGCF</media>	endrecv<	ISUP IAM ACM ANM  CPG(Conference established)				
values:	INVITE 180 Ringing 200 OK INVITE ACK  INVITE(SDP focus) 200 OK INVITE ACK  NOTIFY 1 200 OK NOTIFY  NOTIFY 2 200 OK NOTIFY	<pre><media 1'<="" id="1'&lt;/th&gt;&lt;th&gt;endrecv&lt;&lt;/th&gt;&lt;th&gt;ISUP IAM ACM ANM  CPG(Conference established)  CPG(isolated)&lt;/th&gt;&lt;/tr&gt;&lt;tr&gt;&lt;th&gt;values:&lt;/th&gt;&lt;th&gt;INVITE 180 Ringing 200 OK INVITE ACK  INVITE(SDP focus) 200 OK INVITE ACK  NOTIFY 1 200 OK NOTIFY  NOTIFY 2 200 OK NOTIFY  NOTIFY 3&lt;/th&gt;&lt;th&gt;&lt;pre&gt;&lt;/th&gt;&lt;th&gt;endrecv&lt;&lt;/th&gt;&lt;th&gt;ISUP IAM ACM ANM  CPG(Conference established)&lt;/th&gt;&lt;/tr&gt;&lt;tr&gt;&lt;th&gt;values:&lt;/th&gt;&lt;th&gt;INVITE 180 Ringing 200 OK INVITE ACK  INVITE(SDP focus) 200 OK INVITE ACK  NOTIFY 1 200 OK NOTIFY  NOTIFY 2 200 OK NOTIFY&lt;/th&gt;&lt;th&gt;&lt;pre&gt;&lt;media id=" th=""><th>endrecv&lt;</th><th>ISUP IAM ACM ANM  CPG(Conference established)  CPG(isolated)</th></media></pre>	endrecv<	ISUP IAM ACM ANM  CPG(Conference established)  CPG(isolated)				
values:	INVITE 180 Ringing 200 OK INVITE ACK  INVITE(SDP focus) 200 OK INVITE ACK  NOTIFY 1 200 OK NOTIFY  NOTIFY 2 200 OK NOTIFY  NOTIFY 3 200 OK NOTIFY	<media <status="" id="1">si  MGCF</media>	endrecv<	ISUP IAM ACM ANM  CPG(Conference established)  CPG(isolated)  CPG(reattached)				
values:	INVITE 180 Ringing 200 OK INVITE ACK  INVITE(SDP focus) 200 OK INVITE ACK  NOTIFY 1 200 OK NOTIFY  NOTIFY 2 200 OK NOTIFY  NOTIFY 3 200 OK NOTIFY  BYE	<media <status="" id="1">si  MGCF</media>	endrecv<	ISUP IAM ACM ANM  CPG(Conference established)  CPG(isolated)  CPG(reattached)  REL				
values:	INVITE 180 Ringing 200 OK INVITE ACK  INVITE(SDP focus) 200 OK INVITE ACK  NOTIFY 1 200 OK NOTIFY  NOTIFY 2 200 OK NOTIFY  NOTIFY 3 200 OK NOTIFY	<media <status="" id="1">si  MGCF</media>	endrecv<	ISUP IAM ACM ANM  CPG(Conference established)  CPG(isolated)  CPG(reattached)				

TP504010	SIP reference: RFC 3261 [6]	F	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".</endpoint-type>				
SIP Parameter	NOTIFY 1: see test case 504005				
values:	NOTIFY 2: Event contains <b>conference</b> ; Subscription-State contains <b>active</b> application/conference-info+xml: <conference-info></conference-info>				
	entity=conference URI s <conference-state> <user-count>3<th></th><th></th></user-count></conference-state>				
	<users></users>	Journ 7	Procent		
	<user entity="SIPx" th="" u<=""><th></th><th></th></user>				
	<endpoint entity="&lt;br"><status><b>on-</b>h</status></endpoint>				
	<joining-meth <media full'<="" id="1&lt;/th&gt;&lt;th&gt;od&gt;dialed&lt;/th&gt;&lt;th&gt;d-out&lt;/ joining-method&gt;&lt;/th&gt;&lt;/tr&gt;&lt;tr&gt;&lt;th&gt;&lt;/th&gt;&lt;th&gt;&lt;status&gt;s&lt;/th&gt;&lt;th&gt;&lt;/th&gt;&lt;th&gt;&lt;/th&gt;&lt;/tr&gt;&lt;tr&gt;&lt;th&gt;&lt;/th&gt;&lt;th&gt;NOTIFY 3: Event contains &lt;b&gt;conference&lt;/b&gt;; Su application/conference-info+xm&lt;/th&gt;&lt;th&gt;&lt;/th&gt;&lt;th&gt;n-State contains &lt;b&gt;active&lt;/b&gt;&lt;/th&gt;&lt;/tr&gt;&lt;tr&gt;&lt;th&gt;&lt;/th&gt;&lt;th&gt;&lt;conference-info&gt;&lt;/th&gt;&lt;th&gt;&lt;/th&gt;&lt;th&gt;&lt;/th&gt;&lt;/tr&gt;&lt;tr&gt;&lt;th&gt;&lt;/th&gt;&lt;th&gt;entity=conference URI s&lt;br&gt;&lt;conference-state&gt;&lt;/th&gt;&lt;th&gt;tate=" th=""><th>version="x"</th></media></joining-meth 	version="x"			
	<user-count>3<th>r-count&gt; i</th><th>f present</th></user-count>	r-count> i	f present		
	<users> <user entity="SIPx" th="" u<=""><th>DI etato-"</th><th>'frill"</th></user></users>	DI etato-"	'frill"		
	<pre><endpoint entity="&lt;/pre"></endpoint></pre>				
	<status>coni</status>				
	<pre></pre> <pre></pre> <pre></pre> <pre></pre> <pre></pre> <pre>media id="1</pre>		d-out joining-method		
	<status>s</status>	endrecv<	/status>		
ISUP Parameter values:					
Comments:	SIP MGCF		ISUP		
	INVITE → 180 Ringing ←	<b>→</b>	IAM ACM		
	200 OK INVITE	<b>←</b>	ANM		
	ACK →				
	INVITE(SDP focus) →				
	200 OK INVITE ←				
	ACK →				
	NOTIFY 1 →	<b>→</b>	CPG(Conference established)		
	200 OK NOTIFY ← NOTIFY 2 →	<b>→</b>	CPG(other party isolated)		
	200 OK NOTIFY		or oformor party isolated)		
	NOTIFY 3 → 200 OK NOTIFY ←	<b>→</b>	CPG(other party reattached)		
	BYE ←	<b>←</b>	REL		
	200 OK BYE →	<b>→</b>	RLC		

TP504011	SIP reference: RFC 3261 [6]		NGN reference: ES 283 027 [1], clause 7.5.6		
TSS reference:	SIP-ISUP/SS/CONF/	1			
SIP selection	PICS 1/1				
criteria:					
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".</endpoint-type>				
SIP Parameter values:	<pre></pre>	al:  atate="full r-count>  RI state= endpoint nected aod>diale endrecv abscriptio al:  atate="full r-count>  RI state= endpoint onnected onnected	"version="x"  if present  "full" t SIPx URI status> d-out joining-method c/status> n-State contains active  "version="x"  if present  "full" t SIPx URI		
		on-metho	d>departed <disconnection-< th=""></disconnection-<>		
IOUR T	<status>s</status>	endrecv<			
ISUP Parameter					
values: Comments:	SIP MGCF	•	ISUP		
Comments.	INVITE	<b>→</b> ← ←	IAM ACM ANM		
	NOTIFY 1 → 200 OK NOTIFY ←	<b>→</b>	CPG(Conference established)		
	NOTIFY 2 → 200 OK NOTIFY ←	<b>→</b>	CPG(other party added)		
	NOTIFY 3 → 200 OK NOTIFY ←	<b>→</b>	CPG(other party disconnected)		
	BYE	<b>←</b> <b>→</b>	REL 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 map	ped into "otl	ner party added"		
	Upon the receipt of a conference informati information is not mapped to the PSTN sid				
SIP Parameter values:	information is not mapped to the PSTN side. No NOTIFY is sent to the ISDN user.  NOTIFY 1: see test case 504005  NOTIFY 2: Event contains conference; Subscription-State contains active application/conference-info+xml:				
ISUP Parameter					
values:	SIP MG	CF	ISUP		
Comments:	INVITE  180 Ringing  200 OK INVITE  INVITE(SDP focus)  200 OK INVITE  ACK  →	→ ← ←	IAM ACM ANM		
	NOTIFY 1 → 200 OK NOTIFY ←	<b>→</b>	CPG(Conference established)		
	NOTIFY 2 200 OK NOTIFY ←	<b>→</b>	CPG(other party added)		
	BYE ← 200 OK BYE →	<b>←</b> →	REL RLC		

TP504013	SIP reference: RFC 3261	[6]	l	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 pos-	sible call is es	tablishe	ed
	Ensure that a REFER request recrejected with 403 Forbidden. The			
SIP Parameter	REFER: Request URI contained th			
values:	Refer-To contains the UR Referred-By contains SIP			ite
ISUP Parameter				
values:				
Comments:	SIP	MGCF		ISUP
	INVITE -	•	<b>→</b>	IAM
	180 Ringing ←	•	<b>←</b>	ACM
	200 OK INVITE ←	•	<b>←</b>	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	REFER: Request URI contained the conferen		
values:	Refer-To contains the URI of ISUPx,		
	Referred-By contains SIP or tel URI of	of SIPx	
ISUP Parameter			
values:			
Comments:	SIP MGC	F ISUP	
	REFER →		
	403 Forbidden ←		
	ACK →		

# 6.3.1.5 Three Party service (3PTY)

TP505001	SIP reference: RFC 3261 [6]		F	NGN reference: S 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  Conference established  Conference disconnected was received due to the 3PTY supplementary service.  The media stream is set to:  sendrecv  no change				
SIP Parameter	SDP: a= sendonly				
values:	SDP: a= sendrecv				
ISUP Parameter values:	CPG: notification = remote hold CPG: Generic notification = Conference	ootoblic	ahad		
Comments:		SUT	sneu	ISUP	
Comments.	INVITE -	301	<b>→</b>	IAM	
	180 Ringing		ŕ	ACM	
	Ringing tone		•	7.OW	
	200 OK INVITE		<b>←</b>	ANM	
	ACK →		-	AUTO	
		onversa	tion		
	INVITE(sendonly)	),,, <b>,</b>		CPG(hold)	
	200 OK INVITE(recvonly) →			· · · · (· · · · · · )	
	ACK ←				
	INVITE(sendrecv) ← 200 OK INVITE(sendrecv) → ACK ←		<b>←</b>	CPG(Conference established)	
	Co	onversa			
				CPG(Conference disconnected)	
		onversa	ition		
	BYE →		<b>→</b>	REL	
	200 OK BYE ←			RLC	

TP505002	SIP reference: RFC 3261 [6]		NGN reference: ES 283 027 [1], clause 7.4.15 T Recommendation 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 C service, the Generic notification indication of the SIF No mapping, no disrupting the SIF	ator with the valu	
SIP Parameter	No mapping		
values:			
ISUP Parameter	CPG: Generic notification = Confere		
values:	CPG: Generic notification = Confere		
Comments:	SIP	SUT	ISUP
	INVITE -	=	7, 441
	180 Ringing ←	•	- ACM
	Ringing tone		
	INVITE(sendonly)	•	CPG(hold)
	200 OK INVITE(recvonly)		
	ACK ←		
		Conversation	CPG(Conference established)
		Conversation	CPG(Conference disconnected)
	BYE →	Conversation	REL
	200 OK BYE		
L	ZOO ON DIE		· ILLO

TP504003	SIP reference: RFC	3261 [6]	E	NGN reference: S 283 027 [1], clause 7.4.15		
TSS reference:	SIP-ISUP/SS/3PTY					
SIP selection criteria:	PICS 1/1					
ISUP selection criteria:						
Test purpose:		ence informations, the MGCF st	n documen	nference established" t with the <conference-state-type> CPG message to the CS side with a</conference-state-type>		
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 <endpoint="" <status="" entity="endpoint" isupx="" state="full" uri="">connected <joining-method>dialed-out</joining-method> <media <status="" id="1">sendrecv</media></user-entity=isupx></users></conference-state></conference-info>					
ISUP Parameter						
values: Comments:	SIP INVITE 180 Ringing 200 OK INVITE ACK INVITE(sendonly) 200 OK INVITE(recvonly) ACK	MG	CF	ISUP IAM ACM ANM  CPG(hold)		
	INVITE(SDP focus) 200 OK INVITE ACK NOTIFY 1 BYE 200 OK BYE	→ ← → ←	→ ← →	CPG(Conference established) REL RLC		

# 6.3.1.6 Connected line identification (COL)

TP506001				SUP reference:
			[2	?], 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 nu	ımber not supporte	d	
	Ensure that the SUT, if a signalling procedure. The			NM, does not disrupt the SIP o any SIP message.
SIP Parameter values:				
ISUP Parameter values:	ANM: Connected number	er Parameter		
Comments:	SIP		MGCF	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM
	200 OK INVITE	<b>←</b>	<b>+</b>	ANM
	ACK	<b>→</b>		
		Conv	versation	
	BYE	<b>→</b>	→	REL
	200 OK BYE	<del>(</del>	<b>+</b>	RLC

TP506002			UP reference: uses 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 allo	wed no additional	I connected number received	
rest purpose.	Oormooled namber national, presentation and	wea, no additional	connected namber received	
	Ensure that the SUT, on receipt of an ANM me	essage with a		
	Connected number parameter coded			
	Address presentation restricted parameter = p	resentation allowe	ed	
	Nature of address indicator = national number			
	Numbering plan indicator = ISDN/Telephony r	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	200 OK INVITE: P-Asserted-Identity header fi	eld Tel URL conta	ining an URI in the format	
values:	"+"CC+NDC+SN			
ISUP Parameter	ANM;			
values:	Connected number parameter			
	Address presentation restricted parameter = '0	00'B		
	Nature of address indicator = '0000011'B			
	Numbering plan indicator = '001'B			
	Screening indicator = Network provided Address signals = derived from the P-Asserter	d Idontity		
	Generic number parameter not present	u-lueritity		
Comments:	• •	GCF	ISUP	
	INVITE →	→	IAM	
	180 Ringing ←	<b>É</b>	ACM	
	200 OK INVITE	÷	ANM	
	ACK →	-		
		rsation		
	BYE →	→	REL	
	200 OK BYE ←	<del>-</del>	RLC	
	1=00 0=12			

TP506003				GN 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 internations received	al, presentatio	n allowed, no addi	itional connected number	
	Ensure that the SUT, on receip	t of an ANM n	nessage with a		
	Connected number parameter				
	Address presentation restricted			red	
	Nature of address indicator = international number  Numbering plan indicator = ISDN/Telephony numbering plan				
	Screening indicator = Network		numbering plan		
	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 NDC+ SN as received in the	e connected n	umber in the ANM		
SIP Parameter values:	200 OK INVITE: P-Asserted-Ide	entity header f	ield Tel URL conta	aining an URI in the format	
ISUP Parameter	ANM;				
values:	Connected number paramete	er			
	Address presentation restricted		'00'B		
	Nature of address indicator = 0				
	Numbering plan indicator = '00'				
	Screening indicator = Network provided				
	Address signals = PIXIT  Generic number parameter n	ot propont			
Comments:	SIP		MGCF	ISUP	
Comments.	INVITE	<b>→</b>		IAM	
	180 Ringing	<del>-</del>	<del>-</del>	ACM	
	200 OK INVITE	<del>-</del>	<del>-</del>	ANM	
	ACK	<b>•</b>	•	, vi aini	
	Conversation				
	BYE	→ Conv	->	REL	
	200 OK BYE	÷	É	RLC	
		-	•		

TP506004		ISUP reference: [14], clauses 7.4.2.2 and 7	7.5.2		
TSS reference:	SIP-ISUP/SS/COL/	[14], Clauses 7.4.2.2 and 7	.5.2		
SIP selection	PICS 5/22 AND NOT PICS 13/1				
criteria:					
ISUP selection					
criteria:					
Test purpose:	Connected number national, presenta	tion allowed, additional connected number i	received		
	Francisco de et de a OUT an accesint et an	A N.I.M			
	Ensure that the SUT, on receipt of an ANM message with a				
	Connected number parameter code	d			
	Address presentation restricted param				
	Nature of address indicator = national				
	Numbering plan indicator = ISDN/Tele				
	Screening indicator = Network provide				
	Address signals in the format: NDC+S	IN .			
	Generic number parameter,				
	Number Qualifier Indicator "Additional	connected number"			
	Address presentation restricted param				
	Nature of address indicator = national				
	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				
	D. Accorted Identity bender field containing a LIPI with an identity in the format "" " CC I				
	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 wi "+".				
	The additional connected number				
SIP Parameter		eader field Tel URL containing an URI in th	e format		
values:	"+"CC+NDC+SN				
ISUP Parameter	ANM;				
values:	Connected number parameter	otor - '00'B			
	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				
Comments:	SIP	MGCF ISUP			
	INVITE -	→ IAM			
	180 Ringing ←	<b>←</b> ACM			
	200 OK INVITE	<b>←</b> ANM			
	ACK →				
	DVE	Conversation			
	BYE 200 OK BYE	→ REL			
	200 OK BYE ←	<b>←</b> RLC			

TP506005			IS	SUP reference:	
			[14], cla	uses 7.4.2.2 and 7.5.2	
TSS reference:	SIP-ISUP/SS/COL/				
SIP selection criteria:	PICS 5/22 AND NOT PI	CS 13/1			
ISUP selection criteria:					
Test purpose:	Connected number inter received	rnational, presentatio	on allowed, addition	nal connected number	
	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 in the connected number in the ANM. The additional connected number is not interworked				
SIP Parameter				aining an URI in the format	
values:	"+"CC+NDC+SN	, , , , , , , , , , , , , , , , , , , ,		g	
ISUP Parameter	ANM;				
values:	Connected number paraddress presentation re Nature of address indicated Numbering plan indicator = Net Address signals = PIXIT Generic number param Number Qualifier Indicated Address presentation re Nature of address indicated Numbering plan indicated Screening indicator = '00 Address signals = PIXIT	stricted parameter = ator = "0000100'B ar = '001'B etwork provided  neter for "00000101"B stricted parameter = ator = '0000100'B ar = '001'B D'B	'00'B		
Comments:	SIP		MGCF	ISUP	
	INVITE	<b>→</b>	<b>→</b>	IAM	
	180 Ringing	<b>←</b>	<b>←</b>	ACM	
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM	
	ACK	<b>→</b>			
	DVE		ersation `	DEL	
	BYE	<b>→</b>	<b>→</b>	REL	
	200 OK BYE	+	+	RLC	

TP506006		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, no additional connected number received		
	Ensure that the SUT, on receipt of an ANM m	essage 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		
SIP Parameter	200 OK INVITE: P-Asserted-Identity header fi	eld Tel URL containing an URI in the format	
values:	"+"CC+NDC+SN		
ISUP Parameter	ANM;		
values:	Connected number parameter  Address presentation restricted parameter = ' Nature of address indicator = '0000011'B  Numbering plan indicator = '001'B  Screening indicator = Network provided  Address signals = PIXIT  Generic number parameter not present		
Comments:	SIP	GCF ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →	_	
		ersation	
	BYE -	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

TP506007		ISUP reference: [14], clauses 7.4.2 and 7.5.2		
TSS reference:	SIP-ISUP/SS/COL/	[14], 0144303 7.4.2 4114 7.3.2		
SIP selection	PICS 5/22 AND NOT PICS 13/1			
criteria:	1100 0/22 / 1110 1101 1100 10/1			
ISUP selection				
criteria:				
Test purpose:	Connected number international, presentation received	restricted, no additional connected number		
	Ensure that the SUT, on receipt of an ANM message with a			
	Connected number parameter coded			
	Address presentation restricted parameter = p			
	Nature of address indicator = international nu			
	Numbering plan indicator = ISDN/Telephony r	numbering plan		
	Screening indicator = Network provided	Y. CNI		
	Address signals in the format: PIXIT CC+NDC	7+3N		
	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 in the connected number in the ANM. a Privacy header is inserted with the value "id" or the value "id" is added to a existence Privacy header			
SIP Parameter	200 OK INVITE: P-Asserted-Identity header fi	eld Tel URL containing an URI in the format		
values:	"+"CC+NDC+SN			
ISUP Parameter	ANM;			
values:	Connected number parameter			
	Address presentation restricted parameter = '(	J1'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:		GCF ISUP		
	INVITE ->	→ IAM		
	180 Ringing ←	<b>←</b> ACM		
	200 OK INVITE ←	<b>←</b> ANM		
	ACK →			
	Conve	ersation		
	BYE →	→ REL		
	200 OK BYE ←	<b>←</b> RLC		

TP506008		IS	SUP reference:		
		[14], cla	auses 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 re	stricted, additional	connected number received		
	Ensure that the SUT, on receipt of an ANM r	nessage with a			
	Connected number parameter coded				
	Address presentation restricted parameter =	presentation restri	icted		
	Nature of address indicator = national number				
	Numbering plan indicator = ISDN/Telephony	numbering plan			
	Screening indicator = Network provided Address signals in the format: PIXIT NDC+S	NI			
	Address signals in the format. I fair NDC+3	IN			
	Generic number parameter,				
	Number Qualifier Indicator "Additional conne				
	Address presentation restricted parameter =		icted		
	Nature of address indicator = national number Numbering plan indicator = ISDN/Telephony				
	Screening indicator = user provided, not veri				
	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 constru	uct E.164 number i	in URI. Prefix number with		
	"+".	a "id" or the value	"id" is added to a evictoria		
	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	200 OK INVITE: P-Asserted-Identity header	field Tel URL conta	aining an URI in the format		
values:	"+"CC+NDC+SN				
ISUP Parameter values:	ANM; Connected number parameter				
values.	Address presentation restricted parameter =	'01'B			
	Nature of address indicator = '0000011'B	0.15			
	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				
Comments:	Address signals = PIXIT	MGCF	ISUP		
Comments.	INVITE +	wiger →	IAM		
	180 Ringing	÷	ACM		
	200 OK INVITE	÷	ANM		
	ACK →				
		ersation			
	BYE →	<b>→</b>	REL		
	200 OK BYE <b>←</b>	+	RLC		

TP506009		ISUP reference: [14], clauses 7.4.2 and 7.5.2		
TSS reference:	SIP-ISUP/SS/COL/	4/		
SIP selection	PICS 5/22 AND NOT PICS 13/1			
criteria:				
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			
SIP Parameter	P-Asserted-Identity header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received in the connected number in the ANM 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  200 OK INVITE: P-Asserted-Identity header field Tel URL containing an URI in the format			
values:	"+"CC+NDC+SN	meta i en a contamining an en a minimat		
ISUP Parameter	ANM;			
values:	Connected number parameter  Address presentation restricted parameter = 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 = Nature of address indicator = '0000100'B  Numbering plan indicator = '001'B  Screening indicator = '00'B  Address signals = PIXIT	: '001B		
Comments:		MGCF ISUP		
	INVITE → 180 Ringing ←	→ IAM ← ACM		
	180 Ringing ← 200 OK INVITE ←	← ANM		
	ACK	AINIVI		
		versation		
	BYE →	→ REL		
	200 OK BYE	← RLC		

TP506010			IS	SUP reference:
			[14], cla	uses 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:	IAM connected line request in	diaatian ia aant		
Test purpose:	IAM connected line request in	aication is sent		
	Ensure that a optional forward set to "requested" is contained Supported header equal to "from the contained by the contained	d in the sent IAN om-change".		
SIP Parameter	INVITE: Supported: "from-cha	nge"		
values:				
ISUP Parameter	IAM: oFCi Connected line identity request indicator is set to "requested"			
values:				
Comments:	SIP	N	IGCF	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
		Conv	ersation	
	BYE	<b>→</b>	<b>→</b>	REL
	200 OK BYE	+	+	RLC

TP506011		ISUP reference: [14], clauses 7.4.2 and 7.5.2			
TSS reference:	SIP-ISUP/SS/COL/				
SIP selection	PICS 5/22 AND PICS 13/1				
criteria:					
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", "presentation allowed" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag.				
	Ensure that the From header in the received I	JPDATE (after 200 OK INVITE) contains the			
		CC-NDC-SN". The UPDATE does not contain			
217.7	the Privacy value 'header'.				
SIP Parameter	200 OK INVITE: Supported: "from-change"				
values:		from the Connected number			
IOUD Danamatan	UPDATE: From header contains the Generic				
ISUP Parameter values:	IAM: oFCi Connected line identity request ind ANM:	icator is set to "requested"			
values:	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 = p	presentation allowed			
	Nature of address indicator = national numbe				
	Numbering plan indicator = ISDN/Telephony i	numbering plan			
	Screening indicator = Network provided				
	Address signals = PIXIT				
Comments:		IGCF ISUP			
	INVITE →	→ IAM			
	180 Ringing ←	<b>←</b> ACM			
	200 OK INVITE ←	<b>←</b> ANM			
	ACK →				
	UPDATE <b>←</b>				
	200 OK UPDATE →				
		ersation			
	BYE →	→ REL			
	200 OK BYE ←	<b>←</b> RLC			

TP506012		ISUP reference: [14], clauses 7.4.2 and 7.5.2			
TSS reference:	SIP-ISUP/SS/COL/				
SIP selection	PICS 5/22 AND PICS 13/1				
criteria:					
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" "presentation allowed" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag.				
	Ensure that the From header in the received l	JPDATE (after 200 OK INVITE) contains the			
		CC-NDC-SN". The UPDATE does not contain			
	the Privacy value 'header'.				
SIP Parameter	200 OK INVITE: Supported: "from-change"				
values:		from the Connected number			
IOUD D	UPDATE: From header contains the Generic				
ISUP Parameter	IAM: oFCi Connected line identity request ind	icator is set to "requested"			
values:	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 = 15DN/Telephony numbering plan   Screening indicator = user provided, not verified				
	Address signals = PIXIT				
	Connected number parameter				
	Address presentation restricted parameter = p	presentation allowed			
	Nature of address indicator = international nu				
	Numbering plan indicator = ISDN/Telephony r				
	Screening indicator = Network provided				
	Address signals = PIXIT				
Comments:		GCF ISUP			
	INVITE →	→ IAM			
	180 Ringing ←	<b>←</b> ACM			
	200 OK INVITE ←	<b>←</b> ANM			
	ACK →				
	UPDATE <b>←</b>				
	200 OK UPDATE →				
		ersation			
	BYE →	→ REL			
	200 OK BYE ←	<b>←</b> 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" "presentation restricted" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag.  Ensure that the From header in the received UPDATE (after 200 OK INVITE) contains the Additional connected number in the format +"CC-NDC-SN". The UPDATE contains the Privacy value 'header'.			
SIP Parameter	200 OK INVITE: Supported: "from-change"			
values:	P-Asserted-Identity derived	from the Connected number in the format		
	"+"CC+NDC+SN Privacy: id UPDATE: From header contains the Generic	y wash on its the formest II : IICC : NIDC : CNI		
	Privacy: header	number in the format + CC+NDC+SN		
ISUP Parameter	IAM: oFCi Connected line identity request inc	licator is set to "requested"		
values:	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			
	Screening indicator = Network provided			
	Address signals = PIXIT			
Comments:	1	IGCF ISUP		
	INVITE -	→ IAM		
	180 Ringing	← ACM		
	200 OK INVITE ← ACK →	<b>←</b> ANM		
	ACK →			
	UPDATE <b>←</b>			
	200 OK UPDATE →			
		ersation		
	BYE →	→ REL		
	200 OK BYE ←	<b>←</b> RLC		

TP506014		ISUP reference: [14], clauses 7.4.2 and 7.5.2		
TSS reference:	SIP-ISUP/SS/COL/	[11], oldacoo 11112 alla 11012		
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" "presentation restricted" is included then a 200 OK INVITE is sent and the Supported header contains the "from-change" tag.  Ensure that the From header in the received UPDATE (after 200 OK INVITE) contains the Additional connected number in the format +"CC-NDC-SN". The UPDATE contains the Privacy value 'header'.			
SIP Parameter	200 OK INVITE: Supported: "from-change"			
values:	P-Asserted-Identity derived	from the Connected number in the format		
	"+"CC+NDC+SN Privacy: i	d		
	Privacy: header	number in the format + CC+NDC+SN		
ISUP Parameter	IAM: oFCi Connected line identity request inc	dicator is set to "requested"		
values:	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 = ISDIN/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 nu			
	Numbering plan indicator = ISDN/Telephony	numbering plan		
	Screening indicator = Network provided Address signals = PIXIT			
Comments:		MGCF ISUP		
	INVITE →	→ IAM		
	180 Ringing ←	<b>←</b> ACM		
	200 OK INVITE ←	← ANM		
	ACK →			
	UPDATE			
	200 OK UPDATE →			
	Conv	rersation		
	BYE →	→ REL		
	200 OK BYE ←	<b>←</b> RLC		

## 6.3.1.7 Malicious call identification MCID

TP507001	SIP reference	e: RFC 3261 [6]		ISUP reference:
			ES 2	83 027 [1], clause 7.4.4
TSS reference:	SIP-ISUP/SS/MCID/	,		
SIP selection	PICS 9/1			
criteria:				
ISUP selection				
criteria:				
Test purpose:	No interworking MG	CF sends IRS		
				essage. The MCID response
	indicator is set to "M	CID not included". The SI	P signalling p	rocedure is not disrupted.
SIP Parameter	No influence			
values:				
ISUP Parameter	IDR: MCID reques	ited		
values:	IRS: MCID not inc	luded		
Comments:	SIP	SU	T	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
			<b>←</b>	IDR
			<b>→</b>	IRS
	180 Ringing	<b>←</b>	<b>←</b>	ACM
	i sa sangang	Ringing tone		
	200 OK INVITE	+ till gillig tollo	<b>←</b>	ANM
	ACK	÷	•	/ ti vivi
	/ COIX	Conver	eation	
	DVE			DEI
	BYE	<del>(</del>	<b>+</b>	REL
	200 OK BYE	<b>→</b>	<b>→</b>	RLC

TP507002	SIP referen	ce: RFC 3261 [6]		P reference: 27 [1], clause 7.4.4
TSS reference:	SIP-ISUP/SS/MCID	/		
SIP selection	NOT PICS 9/1			
criteria:				
ISUP selection				
criteria:				
Test purpose:	No interworking time	eout T39		
	not disrupted.	Γ if an IDR is received, no	IDR is sent. The S	SIP signalling procedure is
SIP Parameter	No influence			
values:				
ISUP Parameter	IDR: MCID reques	sted		
values:				
Comments:	SIP	SU		
	INVITE	<b>→</b>	→ IAN	•
			← IDF	₹
				T39 timeout
	180 Ringing	<b>←</b>	<b>←</b> AC	M
		Ringing tone		
	200 OK INVITE	<b>←</b>	← AN	M
	ACK	<b>→</b>		
		Convei	rsation	
	BYE	<b>←</b>	← RE	L
	200 OK BYE	<b>→</b>	→ RL	C

TP507003	SIP reference: RF6	C 3261 [6]	ES 2	ISUP reference: 83 027 [1], clause 7.5.9
TSS reference:	SIP-ISUP/SS/MCID/			
SIP selection	PICS 9/1			
criteria:				
ISUP selection				
criteria:				
Test purpose:	Interworking of IDR			
	INFO request. It is a 'mcid'			dicator is set to 1, sends out an questIndicator=1
SIP Parameter	INFO			
values:	xml version="1.0" encod</th <th>ing="utf-8"?&gt;</th> <th></th> <th></th>	ing="utf-8"?>		
	<mcid></mcid>			
	<pre><request>   <mcidrequestindicator></mcidrequestindicator></request></pre>	string -/MaidPague	etIndicator>	
	<tns:holdingindicator>st</tns:holdingindicator>			
		ring viriolainginaloc	21012	
ISUP Parameter	IDR: MCID requested =1			
values:	,			
Comments:	SIP	SU	IT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	INFO	<b>←</b>	<b>←</b>	IDR
	180 Ringing	<b>←</b>	+	ACM
		ging tone		
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
		Conve		
	BYE	<b>←</b>	<b>←</b>	REL
	200 OK BYE	<b>→</b>	<b>→</b>	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 calling party subaddress in the IAM	header in an INVITE is mapped in the			
	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 to "0 0 0" "NSAP (ITU-T Recommendation X.213 [29] and ISO/IEC 8348 [30] Add.2)"				
SIP Parameter	INVITE:				
values:	P-Asserted-Identity: sip: user part; isub= <s< th=""><th>subaddress&gt;@hostportion</th></s<>	subaddress>@hostportion			
ISUP Parameter values:	IAM: ATP(Calling party subaddress)	·			
Comments:	SIP	MGCF ISUP			
	INVITE -	→ IAM			
	100 Trying ←				
	180 Ringing ←	← ACM			
	200 OK INVITE	<b>←</b> ANM			
	ACK →				
	Communication				
	BYE →	→ REL			
	200 OK BYE ←	<b>←</b> RLC			

TP508002	SUB Reference: ES 283 027 [1], clause 7.4.5	S	election criteria: PICS 5/8			
TSS reference:	SIP-ISUP/SS/SUB/					
Preconditions:						
Test purpose:	The isub parameter of the Request URI in an subaddress in the IAM	INVITE is map	ped in the called party			
	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 to "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>@h</subaddress>	nostportion				
ISUP Parameter values:	IAM: ATP(Called party subaddress)					
Comments:	SIP	MGCF	ISUP			
	INVITE -	<b>→</b>	· IAM			
	100 Trying ←					
	180 Ringing ←	+	ACM			
	200 OK INVITE ← ← ANM					
	ACK →					
	Communication					
	BYE →	<b>→</b>	REL			
	200 OK BYE ←	+	RLC			

	SUB Reference:	Selection criteria:			
TP508003	ES 283 027 [1], clause 7.4.5	PICS 5/8			
TSS reference:	SIP-ISUP/SS/SUB/	1100 0/0			
Preconditions:					
Test purpose:	The connected subaddress in the ANM is may Identity header in the 200 OK INVITE	oped in the isub parameter of the P-Asserted-			
	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 to "0 0 0" "NSAP (ITU-T Recommendation X.213 [29] and ISO/IEC 8348 [30] Add.2)"				
SIP Parameter	200 OK INVITE:				
values:	P-Asserted-Identity: sip: user part; isub=<	subaddress>@hostportion			
ISUP Parameter	IAM: oFCi: connected line request	·			
values:	ANM: ATP(Connected subaddress)				
Comments:	SIP	MGCF ISUP			
	INVITE →	→ IAM			
	100 Trying ←				
	180 Ringing ←	← ACM			
	200 OK INVITE ←	<b>←</b> ANM			
	ACK →	- ,			
	Commu				
	BYE →	→ REL			
	200 OK BYE	<b>←</b> RLC			

## 6.3.1.9 Call diversion (CDIV)

TP509001	SIP reference: RF	C 3261 [6]	ES 2	ISUP reference: 83 027 [1], clause 7.4.6
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	OBCI "call diversion may o	occur" in ACM recei	ved no mapp	ing
		indicator is set to "o		version may occur indicator in may occur", the SIP signalling
SIP Parameter	No mapping			
values:				
ISUP Parameter	ACM optional backward ca	all indicator call dive	ersion may c	occur
values:				
Comments:	SIP	SU	IT	ISUP
	INVITE	<b>→</b>	<b>→</b>	17 (17)
			<b>←</b>	ACM
	180 Ringing	<b>←</b>	<b>←</b>	CPG
		Ringin	g tone	
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
		Conver	sation	
	BYE	<b>←</b>	<b>←</b>	REL
	200 OK BYE	<b>→</b>	<b>→</b>	RLC

TP509002	SIP reference: R	FC 3261 [6]	ES 2	ISUP reference: 283 027 [1], clause 7.4.6
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection	NOT PICS 5/12 AND NO	T PICS 5/13 AND NO	OT PICS 5/14	4 AND NOT PICS 5/15
criteria:				
ISUP selection				
criteria:				
Test purpose:	BCI called party status "r	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 <b>Redirection number, call diversion information, redirection number restriction and generic notification set to "Call is diverting"</b> , the SIP signalling procedure is not disrupted (CFU, CFB, Cdi).			
SIP Parameter	No mapping			
values:				
ISUP Parameter		er, Call diversion info	rmation, Red	direction number restriction,
values:	Generic notification			
Comments:	SIP	SU	IT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
			<b>←</b>	ACM
	180 Ringing	<b>←</b>	<b>←</b>	CPG
		Ringin	g tone	
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
		Conver	rsation	
	BYE	<b>←</b>	<b>←</b>	REL
	200 OK BYE	<b>→</b>	<b>→</b>	RLC

TP509003	SIP reference: RFC 3	261 [6]	FS 28	ISUP reference: 33 027 [1], clause 7.4.6
TSS reference:	SIP-ISUP/SS/CDIV/		2020	50 027 [1], clause 7.4.0
SIP selection criteria:	NOT PICS 5/12 AND NOT PIC	CS 5/13 AND NO	OT 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	No mapping			
values:	11 3			
ISUP Parameter	ACM: Called party status "Su	bscriber free"		
values:	CPG: Redirection number, Ca		rmation, Gene	eric notification
Comments:	SIP	SU	ΙΤ	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
			<b>←</b>	ACM
	180 Ringing	<b>←</b>	<b>←</b>	CPG
		Ringin	g tone	
	200 OK INVITE	<b>←</b>	•	ANM
	ACK	<b>→</b>		
		Conver	sation	
	BYE	<b>←</b>	+	REL
	200 OK BYE	<b>→</b>	<b>→</b>	RLC

TP509004	SIP reference: RF	C 3261 [6]	ES 2	ISUP reference: 283 027 [1], clause 7.4.6
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection	NOT PICS 5/12 AND NOT	PICS 5/13 AND N	OT PICS 5/14	4 AND NOT PICS 5/15
criteria:				
ISUP selection				
criteria:				
Test purpose:	Redirection number restric	tion received in Al	VM no mappin	ng .
	Ensure that the SUT if an	ANM is received w	ith redirectio	n number restriction
	parameter, the SIP signal			
SIP Parameter	No mapping	<b>U</b> 1	•	
values:				
ISUP Parameter	ANM: Redirection number	r restriction		
values:				
Comments:	SIP	S	JT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
			<b>←</b>	ACM
	180 Ringing	<b>←</b>	<b>←</b>	CPG
		Ringir	ng tone	
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
		Conve	ersation	
	BYE	<b>←</b>	<b>←</b>	REL
	200 OK BYE	<b>→</b>	<b>→</b>	RLC

TP509005	SIP reference: RFC 3261 [6]		ISUP reference: 3 027 [1], clause 7.5.4		
TSS reference:	SIP-ISUP/SS/CDIV/	20 20	5 027 [1]; 0lad00 7:0.4		
SIP selection	PICS 10/7				
criteria:					
ISUP selection					
criteria:					
Test purpose:	BCI called party status "no indication" in ACM	received, map	ping of Redirection reason.		
	Ensure that the SUT, on receipt of an ACM medicators parameter coded Called party's status indicator = no indication the Call diversion information parameter convolved to the Call diversion information parameter convolved to the Call diversion information parameter convolved to the Call diversion information = "010"B Redirection reason = ISUP_REASON and the Generic notification indicator parameter (PIXIT) received. A 181 Being Forwarded is sent. The Redirection to the History-Info header in the 181 Being Forwarded into the cause-parameter in the thickness of the convolved to the cause-parameter of the history-Info the convolved to the cause-parameter of the history-Info the cause-parameter of the history-parameter of the history-parameter of the history-parameter of the history-parameter of the	ded  neter coded  on number inclorwarded. A Pro user. The re	uded in the ACM is mapped rivacy header field "history" is direction reason is mapped		
SIP Parameter	181 Being Forwarded: History-Info:				
values:	hi-targeted-to-uri served user; index=1, hi-targeted-ti uri diverted to user; cause= <b>Status-Code</b> ; index=1.1				
ISUP Parameter	in-targeted-ti un diverted to user, cause= <b>statu</b>	S-Code, index	=1.1		
values:					
Comments:	SIP SU	T	ISUP		
	INVITE →	<b>→</b>	IAM		
	181 Being Forwarded ←	<b>←</b>	ACM		
	180 Ringing ←	<b>←</b>	CPG(Alerting)		
	200 OK INVITE ←	<b>←</b>	ANM		
	ACK →				
	Commun				
	BYE →	<b>→</b>	REL		
	200 OK BYE ←	<b>←</b>	RLC		

Values for test purposes TP509005					
ISUP Parameter	Derived value of parameter field	SIP component	Value		
Call diversion information			History-Info header		
Redirection reason	ISUP_REASON	Cause Value in History	Cause value		
	unknown '0000'B	Index; cause-param =	404		
	Unconditional '0011'B	"cause" EQUAL Status-	302		
	User Busy '0001'B	Code	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]	1	ISUP reference: 3 027 [1], clause 7.5.4		
TSS reference:	SIP-ISUP/SS/CDIV/				
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ND PICS 5/15			
criteria:					
ISUP selection					
criteria:					
Test purpose:	BCI called party status "subscriber free" in AC	M received, m	apping of Redirection reason.		
	Ensure that the SUT on receipt of an <b>ACM</b> me	essage indicatir	ng a first diversion with the		
	Backward call indicators parameter coded	J			
	Called party's status indicator = subscriber fre	е			
	the Call diversion information parameter co	oded			
	Notification subscription option = "010"B				
	Redirection reason = ISUP_REASON				
	and the <b>Generic notification indicator parar</b> Notification indicator = call is diverting,	neter coded			
	Redirection number (PIXIT) received				
	A 180 Ringing is sent. The Redirection number	er included in th	ne ACM is mapped into the		
	History-Info header in the 180 Ringing. A Priva				
	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	180 Ringing: History-Info:				
values:	hi-targeted-to-uri served user; index=1,				
ISUP Parameter	hi-targeted-ti uri diverted to user; cause= <b>Status-Code</b> ; index=1.1				
values:					
Comments:	SIP SU	JT	ISUP		
	INVITE →	<b>→</b>	IAM		
	180 Ringing ←	<b>←</b>	ACM		
	200 OK INVITE ←	<b>←</b>	ANM		
	ACK →				
	Commu	nication			
	BYE →	<b>→</b>	REL		
	200 OK BYE <b>←</b>	+	RLC		

	Values for test purposes TP509006						
ISUP Parameter	Derived value of parameter field	SIP component	Value				
Call diversion information			History-Info header				
Redirection reason	ISUP_REASON	Cause Value in History	Cause value				
	unknown '0000'B	Index; cause-param =	404				
	Unconditional '0011'B	"cause" EQUAL Status-	302				
	User Busy '0001'B	Code	486				
	No reply '0010'B		408				
	Deflection during alerting		487				
	'0100'B						
	Deflection immediate		480				
	response '0101'B						
	Mobile subscriber not		503				
	reachable						

TP509007	SIP reference: RFC 3261 [6]	_	SUP reference: 3 027 [1], clause 7.5.4
TSS reference:	SIP-ISUP/SS/CDIV/		
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15		
criteria:			
ISUP selection			
criteria:			
Test purpose:	CPG with Event indicator ALERTING received, mapping of Redirection reason.		
SIP Parameter	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 not 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.		
Values:	180 Ringing: History-Info: hi-targeted-to-uri served user; index=1,		
values.	hi-targeted-to-un served user; index=1, hi-targeted-ti uri diverted to user; cause= <b>Status-Code</b> ; index=1.1		
ISUP Parameter		,	
values:			
Comments:	SIP SU		ISUP
	INVITE →	<b>→</b>	IAM
		<b>←</b>	ACM(no indication)
	180 Ringing ←	<del>(</del>	CPG
	200 OK INVITE ←	+	ANM
	ACK →		
	Commu		
	BYE -	<b>→</b>	REL
	200 OK BYE ←	<del>-</del>	RLC

Values for test purposes TP509007			
ISUP Parameter	Derived value of parameter field	SIP component	Value
Call diversion information			History-Info header
Redirection reason	ISUP_REASON	Cause Value in History	Cause value
	unknown '0000'B	Index; cause-param = "cause" EQUAL Status-Code	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	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15		
criteria:			
ISUP selection			
criteria:			
Test purpose:	CPG with Event indicator PROGRESS received, mapping of Redirection reason.		
SIP Parameter	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 not 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.		
values:	181 Being Forwarded: History-Info: hi-targeted-to-uri served user; index=1,		
Valuoo.	hi-targeted-to-dri served dser, index=1, hi-targeted-ti uri diverted to user; cause= <b>Status-Code</b> ; index=1.1		
ISUP Parameter			
values:			
Comments:	SIP SU		
	INVITE -	→ IAM	
	180 Ringing ←	← ACM	
	181 Being Forwarded	← CPG	
	200 OK INVITE ←	← ANM	
	ACK →	alastia.	
	Commun		
	BYE •	→ REL	
	200 OK BYE <b>←</b>	<b>←</b> RLC	

	Values for test purposes TP509008				
ISUP Parameter	Derived value of parameter field	SIP component	Value		
Call diversion information			History-Info header		
Redirection reason	ISUP_REASON	Cause Value in History	Cause value		
	unknown '0000'B	Index; cause-param =	404		
	Unconditional '0011'B	"cause" EQUAL Status-	302		
	User Busy '0001'B	Code	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: 3 027 [1], clause 7.5.4
TSS reference:	SIP-ISUP/SS/CDIV/		
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15		
criteria:			
ISUP selection			
criteria:	DOL Hd m ut t t H t t uH i AOAA		union of No Office Com
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 me Backward call indicators parameter coded	essage indicati	ng a first diversion with the
	Called party's status indicator = no indication		
	the Call diversion information parameter co		
	Notification subscription option = ISUP_NS	0	
	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 priv-value		
	is escaped in the URI identified the diverted to user.		
SIP Parameter	181 Being Forwarded: History-Info:		
values:	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 SU	T	ISUP
	INVITE →	<b>→</b>	IAM
	181 Being Forwarded ←	<b>←</b>	ACM
	180 Ringing ←	<b>←</b>	CPG(Alerting)
	200 OK INVITE ←	<b>←</b>	ANM
	ACK →		
	Commur	nication	
	BYE →	<b>→</b>	REL
	200 OK BYE ←	+	RLC

	Values for test purposes TP509009			
	SIP component History-Info header, priv-value component	Call diversion information <i>Notification</i> 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		

ES 283 027 [1], clause 7.5.4		
AND DIGG 5/15		
AND PICS 5/15		
CM received, mapping of Notification		
nessage indicating a first diversion with the		
ree		
coded		
)		
Redirection reason = unconditional and the <b>Generic notification indicator parameter</b> 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 priv-value is escaped in the URI identified the diverted to user.		
180 Ringing: History-Info: hi-targeted-to-uri served user; index=1,		
hi-targeted-to-dri served dser, index=1, hi-targeted-ti uri diverted to user; cause=302; ?priv-value; index=1.1		
<i>,</i> 1		
SUT ISUP		
→ IAM		
← ACM		
<b>←</b> ANM		
unication		
→ REL		
← RLC		

Values for test purposes TP509010		
	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]	_	SUP reference: 3 027 [1], clause 7.5.4	
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15			
criteria:				
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 me	ssage indicating	a first diversion with the	
	Event information parameter coded			
	Event indicator = ALERTING,			
	the Call diversion information parameter of			
	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 priv-value is escaped in the			
	URI identified the diverted to user.			
SIP Parameter	180 Ringing: History-Info:			
values:	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	UT	ISUP	
	INVITE →	→	IAM	
		<b>←</b>	ACM	
	180 Ringing ←	<b>←</b>	CPG	
	200 OK INVITE ←	<b>←</b>	ANM	
	ACK →			
	Commi	ınication		
	BYE →	<b>→</b>	REL	
	200 OK BYE ←	+	RLC	

	Values for test purposes TP509011			
	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/	20 200 027 [1]; olddoo 11014	
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 receive	ed, mapping of Redirection number restriction.	
	Ensure that the SUT on receipt of a <b>CPG</b> message indicating a first diversion with the <b>Event information parameter</b> 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 priv-value is escaped in the URI identified the diverted to user.		
SIP Parameter	180 Ringing: History-Info:		
values:	hi-targeted-to-uri served user; index=1,		
IOUD Developed	hi-targeted-ti uri diverted to user; cause=302; ?priv-value; index=1.1		
ISUP Parameter values:	ANM: Redirection number restriction		
Comments:	SIP S	UT ISUP	
	INVITE ->	→ IAM	
	181 Being Forwarded ←	<b>←</b> ACM	
	180 Ringing ←	<b>←</b> CPG	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Commu	unication	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

Values for test purposes TP509012			
	Derived escaped SIP priv-value	Derived value of parameter field	
	component		
VA_01	Privacy header field absent or "none"	ISUP_RDIR_RESTR = Presentation allowed, '00'B	
VA_02	Privacy header field "history"	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	6]		ISUP reference:
T00 (			ES 28	3 027 [1], clause 7.5.4
TSS reference:	SIP-ISUP/SS/CDIV/			
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15			
criteria:				
ISUP selection				
criteria:	ANIA received meaning of the Dedi			
Test purpose:	ANM received, mapping of the Redi	rection nui	riber restrictio	n parameter.
	Ensure that the SUT, on receipt of a	n ANM ma	accana with th	9
	Redirection number restriction parar		•	
	a 200 IK INVITE including a History-			
	escaped in the URI identified the div			
	number restriction parameter.		3	
SIP Parameter	200 OK INVITE: History-Info:			
values:	hi-targeted-to-uri served user; index	=1,		
	hi-targeted-ti uri diverted to user; car	use=302; '	?priv-value; in	dex=1.1
ISUP Parameter	ANM: Redirection number restriction	n		
values:				
Comments:	SIP	SU	T	ISUP
	INVITE →		<b>→</b>	IAM
	181 Being Forwarded		+	ACM
	180 Ringing ←		<b>←</b>	CPG
	200 OK INVITE ←		<b>←</b>	ANM
	ACK →			
		Commu		
	BYE →		<b>→</b>	REL
	200 OK BYE ←		+	RLC
Comments:	SIP	SU		ISUP
	INVITE →		<b>→</b>	IAM
	180 Ringing ←		<b>←</b>	ACM
	200 OK INVITE ←		<b>←</b>	ANM
	ACK →			
		Commu		
	BYE →		<b>→</b>	REL
	200 OK BYE		+	RLC

Values for test purposes TP509013			
	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"	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	NOT PICS 1/5 AND PICS 10/6			
criteria:				
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 on receip	ot of an INVITE massage with Cause Value		
	in History Index; cause-param = "cause" EQU			
	header field is absent and with the complete C			
	contained in the URI of first Index entry of His			
	SN. The SUT sends:			
	an IAM message with the Redirection inform	ation parameter coded		
	Redirection counter = 1			
	Redirecting reason = ISUP_RR	adad		
	and the <b>Original called number parameter</b> of Nature of address indicator = national number			
	Numbering plan indicator = ISDN/Telephony numbering plan			
	Address presentation restricted parameter = p			
	Address signals included in the format NDC+S			
SIP Parameter	INVITE: History-Info: hi-targeted-to-uri served			
values:		to user; cause=Status-Code; index=1.1		
ISUP Parameter	IAM: Original called number parameter cod	ed		
values:	Nature of address indicator = national number			
	Numbering plan indicator = ISDN/Telephony r			
	Address presentation restricted parameter = p			
	Address signals userinfo of the hi-targeted-to			
Comments:	SIP SUT ISUP			
	INVITE -	→ IAM		
	180 Ringing	← ACM		
	200 OK INVITE ←	<b>←</b> ANM		
	7.01.	alaatian		
	Commu			
	BYE  200 OK BYF  ←	→ REL ← RIC		
	200 OK BYE ←	<b>←</b> RLC		

	Values for test purposes TP509014				
ISUP Parameter	Derived value of	SIP component	Value		
	parameter field				
IAM		INVITE			
Redirection Information	Redirecting reason ISUP_RR	Cause Value in History Index; cause-param =	Cause value		
	unknown '0000'B	"cause" EQUAL	404		
	Unconditional '0011'B	Status-Code	302		
	User Busy '0001'B	1	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]	-	SUP reference: 3 027 [1], clause 7.5.4
TSS reference:	SIP-ISUP/SS/ CDIV /	E3 200	5 027 [1], clause 7.5.4
SIP selection	31F-130F/33/ CDIV /		
criteria:			
ISUP selection	PICS 1/5 AND PICS 10/6		
criteria:			
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 of an INVITE message with Cause Value in History Index; cause-param = "cause" EQUAL Status-Code defined in the table, the Privacy header is absent and with the complete <b>Original called number parameter</b> contained URI of first Index entry of History-Info header in the format "+" CC NDC SN. The SUT sends		
	an 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 = p		owed
	Address signals included in the format CC+NE		
SIP Parameter	INVITE: History-Info: hi-targeted-to-uri served	user; index=1,	
values:			=Status-Code; index=1.1
ISUP Parameter	IAM: Original called number parameter code		
values:	Nature of address indicator = international nur		
	Numbering plan indicator = ISDN/Telephony n		
	Address presentation restricted parameter = p		owed
0	Address signals userinfo of the hi-targeted-to from index 1		
Comments:	SIP SU		ISUP
	INVITE -	<b>→</b>	IAM
	180 Ringing	<del>(</del>	ACM
	200 OK INVITE	<b>←</b>	ANM
	ACK →	alaatla.	
	Commu	_	DEL
	BYE -	<b>→</b>	REL
	200 OK BYE <b>←</b>	+	RLC

	Values for test purposes TP509015				
ISUP Parameter or IE	Derived value of	SIP component	Value		
	parameter field				
IAM		INVITE			
Redirection Information	Redirecting reason ISUP_RR	Cause Value in History Index; cause-param =	Cause value		
	unknown '0000'B	"cause" EQUAL	404		
	Unconditional '0011'B	Status-Code	302		
	User Busy '0001'B	1	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]		SUP reference:
		ES 283	3 027 [1], clause 7.5.4
TSS reference:	SIP-ISUP/SS/ CDIV /		
SIP selection			
criteria:			
ISUP selection	PICS 10/6		
criteria:			
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, on receipt of an 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 <b>Original called number parameter</b> contained URI of first Index entry of History-Info header in the format "+" CC NDC SN. The SUT sends		
	an IAM message with the Redirection inform Redirection counter = 1	ation paramet	er coded
	Redirecting reason = ISUP_RR and the <b>Original called number parameter</b> coded		
	Address presentation restricted parameter = p		tricted
SIP Parameter	INVITE: History-Info: hi-targeted-to-uri served		
values:	hi-targeted-ti uri diverted		
ISUP Parameter	IAM: Original called number parameter code		-otatas coas, masx-1.1
values:	Numbering plan indicator = ISDN/Telephony n		
	Address presentation restricted parameter = p		tricted
	Address signals userinfo of the hi-targeted-to		
Comments:	SIP SU	Т	ISUP
	INVITE →	<b>→</b>	IAM
	180 Ringing ←	<del>(</del>	ACM
	200 OK INVITE ←	<b>←</b>	ANM
	ACK →		
	Commu	nication	
	BYE →	<b>→</b>	REL
	200 OK BYE ←	+	RLC

Values for test purposes TP509016				
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 value	
	unknown '0000'B	"cause" EQUAL	404	
	Unconditional '0011'B	Status-Code	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 /	LO 203 027 [1], clause 7.3.4	
SIP selection	011 -1001 /00/ 0D1V /		
criteria:			
ISUP selection	NOT PICS 1/5 AND PICS 10/6		
criteria:			
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 on receipt of 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 <b>Redirecting number parameter</b> contained in the hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN. The SUT sends:		
	an IAM message with the Redirection inform	nation parameter coded	
	Redirection counter = 1	·	
	Redirecting reason = ISUP_RR		
	and the Redirecting number parameter code		
	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony numbering plan		
	Address presentation restricted parameter = presentation allowed		
SIP Parameter	Address signals included in the format NDC+S		
values:	INVITE: History-Info: hi-targeted-to-uri served	to user; cause=Status-Code; index=1.1	
ISUP Parameter	IAM: Redirecting number parameter coded	to user, cause=Status-Coue, index=1.1	
values:	Nature of address indicator = national number		
valuos.	Numbering plan indicator = ISDN/Telephony r		
	Address presentation restricted parameter = p		
	Address signals userinfo of the hi-targeted-to		
Comments:	SIP SU		
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	200 OK ÎNVÎTE ←	← ANM	
	ACK →		
	Commu	nication	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

Values for test purposes TP509017				
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 value	
	unknown '0000'B	"cause" EQUAL	404	
	Unconditional '0011'B	Status-Code	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 /	1, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1	
SIP selection			
criteria:			
ISUP selection	PICS 1/5 AND PICS 10/6		
criteria:			
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 recei in History Index; cause-param = "cause" EQU header field is absent and with the complete Fhi-targeted-to-uri of History-Info header in the	AL Status-Code defined in the table, Privacy Redirecting number parameter contained	
	an IAM message with the Redirection inform	nation parameter coded	
	Redirection counter = 1	nation parameter season	
	Redirecting reason = ISUP_RR		
	and the Redirecting number parameter code	ed	
	Nature of address indicator = international nur		
	Numbering plan indicator = ISDN/Telephony numbering plan		
	Address presentation restricted parameter = p		
	Address signals included in the format CC+NI		
SIP Parameter	INVITE: History-Info: hi-targeted-to-uri served		
values:		to user; cause=Status-Code; index=1.1	
ISUP Parameter	IAM: Redirecting number parameter coded		
values:	Nature of address indicator = international nur		
	Numbering plan indicator = ISDN/Telephony r Address presentation restricted parameter = p		
Comments:	Address signals userinfo of the hi-targeted-to from index 1  SIP SUT ISUP		
Comments.	INVITE →	→ IAM	
	180 Ringing	← ACM	
	200 OK INVITE	← ANM	
	ACK →	ZUNIN	
	Commu	nication	
	BYE →	→ REL	
	200 OK BYE ←	← RLC	
	ZOO ON DIE	· NLO	

Values for test purposes TP509018				
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 value	
	unknown '0000'B	"cause" EQUAL	404	
	Unconditional '0011'B	Status-Code	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]		SUP reference: 3 027 [1], clause 7.5.4	
TSS reference:	SIP-ISUP/SS/ CDIV /			
SIP selection				
criteria:				
ISUP selection	PICS 10/6			
criteria:				
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, on receipt of an 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 <b>Redirecting number parameter</b> contained hi-targeted-to-uri of History-Info header in the format "+" CC NDC SN. The SUT sends			
	an IAM message with the Redirection information parameter coded			
	Redirection counter = 1			
	Redirecting reason = ISUP_RR			
	and the Redirecting number parameter coded			
SIP Parameter	Address presentation restricted parameter = presentation restricted			
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	IAM: Redirecting number parameter coded			
values:	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 SU	T	ISUP	
	INVITE →	<b>→</b>	IAM	
	180 Ringing ←	<b>←</b>	ACM	
	200 OK INVITE ←	<b>←</b>	ANM	
	ACK →			
	Commur	nication		
	BYE →	→	REL	
	200 OK BYE ←	+	RLC	

Values for test purposes TP509019			
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 value
	unknown '0000'B	"cause" EQUAL	404
	Unconditional '0011'B	Status-Code	302
	User Busy '0001'B	1	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 /	20 200 021 [1], olduse 1.0.4		
SIP selection	011 1001 7007 0511 7			
criteria:				
ISUP selection	PICS 10/6			
criteria:				
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 <b>Original called number parameter</b> contained in the URI of first Index entry of History-Info header, in the format "+" CC NDC SN the Privacy header field is absent. The <b>Redirecting number parameter</b> 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 <b>Redirecting number parameter</b> 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	INVITE: History-Info: hi-targeted-to-uri served user; index=1,			
values:	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	IAM: Original called number parameter coded			
values:	Nature of address indicator = national number			
	Numbering plan indicator = ISDN/Telephony r			
	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 r			
	Address presentation restricted parameter = presentation allowed			
	Address signals userinfo of the hi-targeted-to	from index 1.1		
Comments:	SIP SU			
	INVITE →	→ IAM		
	180 Ringing ←	<b>←</b> ACM		
	200 OK INVITE ←	<b>←</b> ANM		
	ACK →			
	Commu			
	BYE →	→ REL		
	200 OK BYE ←	← RLC		

Values for test purposes TP509020			
ISUP Parameter or IE	Derived value of	SIP component	Value
	parameter field		
IAM		INVITE	
Redirection Information	Redirecting reason	Cause Value in History	Cause value
	ISUP_RR	Index; cause-param =	
	unknown '0000'B	"cause" EQUAL	404
	Unconditional '0011'B	Status-Code	302
	User Busy '0001'B		486
	No reply '0010'B		408
	Deflection during		487
	alerting '0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP509021	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
TSS reference:	SIP-ISUP/SS/ CDIV /	<b>5.</b> 47	
SIP selection			
criteria:			
ISUP selection	PICS 10/6		
criteria:			
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 without 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 <b>Original called number parameter</b> contained the URI of first Index entry of History-Info header in the format "+" CC NDC SN, the Privacy header field is absent. The <b>Redirecting number parameter</b> 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 <b>IAM</b> message with the <b>Redirection information parameter</b> 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 <b>Redirecting number parameter</b> coded		
	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony numbering plan		
	Address presentation restricted parameter = presentation restricted		
OID D	Address signals included		
SIP Parameter	INVITE: History-Info: hi-targeted-to-uri served user; index=1,		
values:	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	IAM: Original called number parameter coded		
values:	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony r		
	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 SL		
	INVITE ->	→ IAM	
	180 Ringing ←	← ACM	
	200 OK INVITE	← ANM	
	ACK →		
	Commu	nication	
	BYE →	→ REL	
	200 OK BYE ←	<b>←</b> RLC	

Table 38: Values for test purposes TP509021

ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information		Cause Value in History	Cause value
	unknown '0000'B	Index; cause-param =	404
	Unconditional '0011'B	"cause" EQUAL	302
	User Busy '0001'B	Status-Code	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	PICS 10/6		
criteria:			
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 <b>Original called number parameter</b> contained the URI of first Index entry of History-Info header in the format "+" CC NDC SN the Privacy value set to "history". The <b>Redirecting number parameter</b> 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		
	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	INVITE: History-Info: hi-targeted-to-uri served user?Privacy=history; index=1,		
values:	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	IAM: Original called number parameter cod		
values:	Nature of address indicator = national number		
	Numbering plan indicator = ISDN/Telephony r		
	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		
Comments:	SIP SU	JT ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	200 OK INVITE ←	← ANM	
	ACK →		
	Commu		
	BYE →	→ REL	
	200 OK BYE	<b>←</b> RLC	

Values for test purposes TP509022			
ISUP Parameter or IE	Derived value of parameter field	SIP component	Value
IAM		INVITE	
Redirection Information		Cause Value in History	Cause value
	unknown '0000'B	Index; cause-param =	404
	Unconditional '0011'B	"cause" EQUAL	302
	User Busy '0001'B	Status-Code	486
	No reply '0010'B		408
	Deflection during alerting '0100'B		487
	Deflection immediate response '0101'B		480
	Mobile subscriber not		503
	reachable		

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	PICS 10/6					
criteria:						
Test purpose:	CDIV performed, the second hi-targeted-to-ur number and the first hi-targeted-to-uri Privacy number Privacy					
	Ensure that the SUT in the Idle state, on recein History Index; cause-param = "cause" EQU the complete <b>Original called number parameter</b> History-Info header in the format "+" CC NDC The <b>Redirecting number parameter</b> is contal History-Info header in the format "+" CC NDC	AL Status-Code defined in the table and with eter contained the URI of first Index entry of SN the Privacy is set to "history". ined in the second hi-targeted-to-uri of				
	Sends an IAM message with the Redirection Redirection counter 2	information parameter coded				
	Redirecting reason = ISUP_RR, the Original called number parameter code	4				
	Nature of address indicator = international nur					
	Numbering plan indicator = ISDN/Telephony r					
	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					
SIP Parameter	Address signals included INVITE: History-Info: hi-targeted-to-uri served	usar2Privacy=history; index=1				
values:		to user C?Privacy=history; cause=302;				
	hi-targeted-ti uri diverted to user D; cause=Status-Code; index=1.1.1					
ISUP Parameter	IAM: Original called number parameter coded					
values:	Nature of address indicator = national number					
	Numbering plan indicator = ISDN/Telephony r					
	Address presentation restricted parameter = p					
	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 r					
	Address presentation restricted parameter = p					
	Address signals userinfo of the hi-targeted-to					
Comments:	SIP SL					
	INVITE →	→ IAM				
	180 Ringing ←	<b>←</b> ACM				
	200 OK INVITE ←	← ANM				
	ACK →					
	Commu					
	BYE →	→ REL				
	200 OK BYE ←	← RLC				

	Values for test purposes TP509023						
ISUP Parameter or IE	Derived value of	SIP component	Value				
	parameter field	-					
IAM		INVITE					
Redirection Information		Cause Value in History	Cause value				
	unknown '0000'B	Index; cause-param =	404				
	Unconditional '0011'B	"cause" EQUAL	302				
	User Busy '0001'B	Status-Code	486				
	No reply '0010'B		408				
	Deflection during		487				
	alerting '0100'B						
	Deflection immediate		480				
	response '0101'B						
	Mobile subscriber not		503				
	reachable						

TP509024	SIP reference: RFC 3261 [6]	ISUP reference:					
11 303024	Sir reference. Ki C 3201 [0]	ES 283 027 [1], clause 7.5.4					
TSS reference:	SIP-ISUP/SS/ CDIV /						
SIP selection	511 1551 755, GB11 7						
criteria:							
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ND PICS 5/15					
criteria:							
Test purpose:	CDIV performed, the third hi-targeted-to-uri is mapped from the latest history entry	sent in the redirecting Redirection counter is					
	Ensure that the SUT in the Idle state, on recei History-Info header with three History entries, into the <b>Original called number parameter</b> ; tinto the <b>Redirecting number parameter</b> Cau "cause" EQUAL Status-Code defined in the ta	the hi-targeted-to-uri of first index is mapped the hi-targeted-to-uri of third index is mapped se Value in History Index; cause-param =					
	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	IAM: Original called number parameter coded						
values:	Nature of address indicator = national number						
	Numbering plan indicator = ISDN/Telephony r						
	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 SU						
	INVITE ->	→ IAM					
	180 Ringing ←	<b>←</b> ACM					
	200 OK INVITE ←	<b>←</b> ANM					
	ACK →						
	Commu	nication					
	BYE →	→ REL					
	200 OK BYE ←	<b>←</b> RLC					

	Values for test purposes TP509024						
ISUP Parameter or IE	Derived value of	SIP component	Value				
	parameter field						
IAM		INVITE					
Redirection Information		Cause Value in History	Cause value				
	unknown '0000'B	Index; cause-param =	404				
	Unconditional '0011'B	"cause" EQUAL	302				
	User Busy '0001'B	Status-Code	486				
	No reply '0010'B		408				
	Deflection during		487				
	alerting '0100'B						
	Deflection immediate		480				
	response '0101'B						
	Mobile subscriber not		503				
	reachable						

TP509025	SIP reference: RFC 3261 [6]	ISUP reference:			
TSS reference:	SIP-ISUP/SS/ CDIV /	ES 283 027 [1], clause 7.5.4			
SIP selection criteria:	311 -1301 /33/ GBIV /				
ISUP selection criteria:	NOT PICS 5/12 AND NOT PICS 5/13 AND NO	OT PICS 5/14 AND NOT PICS 5/15			
Test purpose:	Interworking not supported, session successfu	ul			
	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 related parameters in the IAM and the session setup is not disrupted.				
SIP Parameter	INVITE: History-Info: hi-targeted-to-uri served				
values:		I to user C; cause=Status code; index=1.1 I to user D; cause=Status code; index=1.1.1			
ISUP Parameter	IAM: no mapping				
values:					
Comments:	SIP SU				
	INVITE →	→ IAM			
	180 Ringing ←	← ACM			
	200 OK INVITE ←	<b>←</b> ANM			
	ACK →				
	Commu	nication			
	BYE →	→ REL			
	200 OK BYE ←	← RLC			

# 6.3.1.10 Call waiting (CW)

TP510001	SIP reference: I	RFC 3261 [6]	ES 2	ISUP reference: 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 Wai Ensure that the indication a waiting call" is not inte	on for Call Waiting co	ntained in an <i>i</i>	ACM, Generic notification "call is
SIP Parameter values:				
ISUP Parameter values:	ACM: Generic notificati	on parameter = "Call	is a waiting c	all"
Comments:	SIP	S	UT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b> Ringi	← ng tone	ACM
	200 OK INVITE	←	<b>+</b>	ANM
	ACK	<b>→</b>		
		Conve	ersation	
	BYE	<del>(</del>	<b>←</b>	REL
	200 OK BYE	<b>→</b>	<b>→</b>	RLC

TP510002	SIP reference: RFC 3261 [6]	FS 2	ISUP reference: 83 027 [1], clause 7.4.9
TSS reference:	SIP-ISUP/SS/CW/		03 027 [1], clause 7.4.3
SIP selection	511 1551 75675117		
criteria:			
ISUP selection			
criteria:			
Test purpose:	No mapping of Call Waiting indication in		
	Ensure that the indication for Call Waitin		
	a waiting call", is not interworked in a 18	0 Ringing Respons	se
SIP Parameter	180 Ringing		
values:			
ISUP Parameter	ACM: Called party status "no indication"		
values:	CPG: Generic notification parameter = '		
Comments:	SIP	SUT	ISUP
	INVITE →	<b>→</b>	IAM
		<b>←</b>	ACM
	180 Ringing ←	+	CPG
	R	linging tone	
	200 OK INVITE ←	<b>←</b>	ANM
	ACK →		
	C	onversation	
	BYE ←	<b>←</b>	REL
	200 OK BYE →	<b>→</b>	RLC

# 6.3.1.11 User to user signalling (UUS)

TP511001	SIP reference: RFC 32	61 [6]		ISUP reference: c. Q.1912.5 [32], annex B.21 c. 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 FA with FRJ Ensure that the SUT if a FAR is essential) after call setup, sent is not disrupted.	received with	n an <b>user-to-u</b>	
SIP Parameter values:				
ISUP Parameter	FRJ: User-to-user indicator =	"Service 3 no	t provided"	
values:				
Comments:	SIP	S	UT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b> Ringi	ng tone	ACM
	200 OK INVITE ACK	<b>←</b> →	<b>*</b>	ANM
		Conve	ersation	
			<b>←</b>	FAR
			→	FRJ
			ersation	
	BYE	<b>←</b>	+	REL
	200 OK BYE	<b>→</b>	<b>→</b>	RLC

TP511002	SIP reference: RFC 3	261 [6]		ISUP reference: c. Q.1912.5 [32], annex B.21 IU-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 Ensure that the SUT if a FAR essential) after call setup, the an implicit rejection.	is received with	an user-to-u	FAR ser service 3 request (not not disrupted. No FRJ is sent as
SIP Parameter values:				
ISUP Parameter values:				
Comments:	SIP	SI	JT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b> Bingin	<del>(</del>	
	200 OK INVITE	← Kiligii	ng tone	ANM
	ACK	<b>→</b>	•	ANIVI
	ACK	=	rsation	
		Conve	+	FAR
		Conve	rsation	1741
	BYE	<b>←</b>	+ C	REL
	200 OK BYE	<b>→</b>	<b>→</b>	RLC

TP511003	SIP reference: F	RFC 3261 [6]		ISUP reference: c. Q.1912.5 [32], annex B.21 J-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 in	nplicit request		
SIP Parameter		IAM. The data fie of the User-to-Use	ld of the User-to-	in the INVITE a User-to-user user information is derived from IVITE
values:				
ISUP Parameter values:	IAM: User-to-user inform	nation		
Comments:	SIP		SUT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<b>←</b>	<b>←</b>	ACM
		Ri	nging tone	
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
		Co	nversation	
	BYE	<b>←</b>	<b>←</b>	REL
	200 OK BYE	<b>→</b>	<b>→</b>	RLC

TP511004	SIP reference: RFC 32	61 [6]		IT	ISUP reference: c. Q.1912.5 [32], annex B.21 U-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 respons	е			
	Ensure that the User-to-user in uuidata of an User-to-User hea				CM, ANM or REL is mapped in sponse, final response or BYE
SIP Parameter	18x: User-to-User uuidata				
values:	200: User-to-User uuidata				
	BYE: User-to-User uuidata				
ISUP Parameter	ACM: User-to-user information				
values:	ANM: User-to-user information				
	REL: User-to-user information				
Comments:	SIP		SUT		ISUP
	INVITE	<b>→</b>		_	IAM
	180 Ringing	<b>←</b>		<b>←</b>	ACM
			Ringing tone		
	200 OK INVITE	<b>←</b>		<b>←</b>	ANM
	ACK	<b>→</b>			
			Conversation		
	BYE	<b>←</b>		<b>←</b>	REL
	200 OK BYE	<b>→</b>		<b>→</b>	RLC

# 6.3.1.12 Explicit call transfer (ECT)

TP512001	SIP reference: RFC 3261 [6]		ES 2	ISUP reference: 83 027 [1], clause 7.4.8
TSS reference:	SIP-ISUP/SS/ECT/			
SIP selection				
criteria:	DICC 40/4			
ISUP selection criteria:	PICS 12/1			
Test purpose:	Loop prevention procedure supported, a	a LOP i	esponse "in	sufficient information" is sent
	Ensure that the SUT if a LOP(request) is indication "insufficient information" conti procedure. Ensure that the SUT if a FAC is received procedure.	nue wit	hout disrupt	ing the SIP signalling
SIP Parameter				
values:				
ISUP Parameter	LOP: Response "insufficient information	n"		
values:				10115
Comments:	SIP	SU		ISUP
	INVITE →		<b>→</b>	7 (11)
	180 Ringing ←		<b>←</b>	ACM
		Ringing		
	200 OK INVITE		<b>←</b>	ANM
	ACK →			
		Convers	sation	
			<b>←</b>	LOP
			→	LOP
			<b>←</b>	FAC
		Convers	sation	
	BYE <b>←</b>		<b>←</b>	REL
	200 OK BYE →		→	RLC

TP512002	SIP reference: RFC 32	61 [6]	ES 2	ISUP reference: 83 027 [1], clause 7.4.8
TSS reference:	SIP-ISUP/SS/ECT/			
SIP selection criteria:				
ISUP selection criteria:	NO PICS 12/1			
Test purpose:	Ensure that the SUT if a LOP (raignalling procedure.	request) is rece		· -
	Ensure that the SUT if a FAC is procedure.	received cont	inue without o	disrupting the SIP signalling
SIP Parameter values:				
ISUP Parameter values:				
Comments:	SIP	SU	IT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	180 Ringing	<del>-</del>	+	ACM
		Ringin	g tone	
	200 OK INVITE	<b>←</b>	+	ANM
	ACK	<b>→</b>		
		Conver		
			<b>←</b>	LOP
		•	<b>←</b>	FAC
		Conver		
	BYE	<del>(</del>	<b>←</b>	REL
	200 OK BYE	<b>→</b>	<b>→</b>	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 receive		
	Ensure that the SUT if a REL is received contained a Diagnostic field and the CCBS		
	indicator is coded as CCBS possible:		
	<ul> <li>continue without disrupting the S</li> </ul>	SIP signalling procedure.	
SIP Parameter			
values:			
ISUP Parameter	REL: Cause indicator Diagnostics CCBS poss	sible	
values:			
Comments:	SIP SU	JT 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 28	33 027 [1], clause 7.4.11
TSS reference:	SIP-ISUP/SS/CCNR/			
SIP selection				
criteria:				
ISUP selection				
criteria:				
Test purpose:	CCNR possible in the A	CM received		
	Ensure that the SUT if a	an ACM is received an	d a CCNR Po	essible Indicator is included:
OID D	continue w	vithout disrupting the S	SIP signalling	procedure.
SIP Parameter				
values:				
ISUP Parameter	ACM: CCNR possible in	ndicator CCNR possibl	le	
values:				
Comments:	SIP	SU		ISUP
	INVITE	<b>→</b>	<b>→</b>	I/ 4141
	180 Ringing	<b>←</b>	<b>←</b>	ACM
		Ringin	g tone	
	200 OK INVITE	<b>←</b>	<b>←</b>	ANM
	ACK	<b>→</b>		
		Conve	rsation	
	BYE	<b>←</b>	<b>←</b>	REL
	200 OK BYE	<b>→</b>	<b>→</b>	RLC

### 6.3.1.15 Anonymous Call Rejection (ACR)

TP515001	SIP reference:	RFC 3261 [6]	FS 25	ISUP reference: 33 027 [1], clause 7.4.23
TSS reference:	SIP-ISUP/SS/ACR/			50 027 [1], ciause 7.4.25
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	the call at	a destination user ha		ne ACR supplementary service: value 24 "call rejected due to
SIP Parameter	INVITE: Privacy-head	der = "id"		
values:				nendation Q.850 [5]; cause=24
ISUP Parameter values:	EL: Cause va	alue: 24 "call rejected	I due to ACR s	upplementary service"
Comments:	SIP	5	UT	ISUP
	INVITE	<b>→</b>	<b>→</b>	IAM
	603 Decline	<b>←</b>	<b>←</b>	REL
	ACK	<b>→</b>	<b>→</b>	RLC

# 6.3.1.16 Closed user group (CUG)

TP516001	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 "00"  Ensure that the cougCommunicationIndicators value "00" contained in the INVITE cougs.</cug>		
	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 equal "00". No mapping of <networkindicator> and <cuginterlockbinarycode> into Closed User Group interlock code.</cuginterlockbinarycode></networkindicator></cug></cugcommunicationindicator>		
SIP Parameter	INVITE:		
values:	<pre><cug>     <networkindicator>[PIXIT][PIXIT]</networkindicator></cug></pre>	erlockBinaryCode>	
	<pre><cugcommunicationindicator>00</cugcommunicationindicator></pre>	mmunicationIndicator>	
ISUP Parameter	IAM:		
values:	Optional Forward Call Indicator CUG call indic When optional forward call indicator have to b to "0"		
Comments:	SIP SU	T ISUP	
	INVITE ->	→ IAM	
	180 Ringing ←	← ACM	
	Ringin	g tone	
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conver	rsation	
	BYE ←	<b>←</b> REL	
	200 OK BYE →	→ RLC	

TP516002	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	PICS 5/7		
criteria:			
Test purpose:	Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "01"</cug>		
	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.</cuginterlockbinarycode></networkindicator></cug></cugcommunicationindicator>		
SIP Parameter	INVITE:		
values:	<pre><cug>      <networkindicator>[PIXIT]</networkindicator>      <cuginterlockbinarycode>[PIXIT]</cuginterlockbinarycode>      <cugcommunicationindicator>01</cugcommunicationindicator> </cug></pre>		
ISUP Parameter	IAM:		
values:	Optional Forward Call Indicator CUG call indic When optional forward call indicator have to b to "0"		
Comments:	SIP SU	T ISUP	
	INVITE →	→ IAM	
	180 Ringing ←	← ACM	
	Ringin	g tone	
	200 OK INVITE ←	← ANM	
	ACK →		
	Conver		
	BYE	<b>←</b> REL	
	200 OK BYE →	→ RLC	

TP516003	SIP reference: RFC 3261 [6]	ES 28	ISUP reference: 33 027 [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 receing 10"</cug>	ved INVITE cu	gCommunicationIndicator value
	Ensure that the <cugcommunicationindicate <cug="" a="" body="" call="" forward="" i="" in="" is="" optional="" sent="" xml=""> <networkindicator> is mapped into Network identity and the XML <cug> <cug bina<="" closed="" code="" group="" iam="" interlock="" th="" user=""><th>ndicator - CUG the IAM Closed InterlockBina</th><th>call indicator ="10". The XML d User Group interlock code</th></cug></cug></networkindicator></cugcommunicationindicate>	ndicator - CUG the IAM Closed InterlockBina	call indicator ="10". The XML d User Group interlock code
SIP Parameter	INVITE:		
values:	<cug></cug>		
	<networkindicator>[PIXIT]</networkindicator> <cuginterlockbinarycode>[PIXIT]</cuginterlockbinarycode>		
	<pre><cugcommunicationindicator>10</cugcommunicationindicator></pre>		
ISUP Parameter	IAM:		
values:	Optional Forward Call Indicator CUG call indicator = "10"		
	Closed User Group interlock code		
	Binary code derived from INVITE XML		
0	Network identity derived from INVITE	•	
Comments:		SUT	ISUP
	INVITE -	<b>→</b>	17 (14)
	180 Ringing ←	<b>+</b>	ACM
	_	ing tone	0.N.I.N.4
	200 OK INVITE ← ACK →	<b>←</b>	ANM
	7.0.1	ersation	
	BYE Conv	ersation	REL
	200 OK BYE →	7	
	ZUU UN DIE 7	<b>→</b>	KLU

TP516004	SIP reference: RFC 3261 [6]	ES 28	ISUP reference: 3 027 [1], clause 7.4.1.1
TSS reference:	SIP-ISUP/SS/CUG/		
SIP selection			
criteria:			
ISUP selection	PICS 5/7		
criteria:			
Test purpose:	Mapping of <cug> XML element in the received INVITE cugCommunicationIndicator value "11"</cug>		
	Ensure that the <cugcommunicationindicato <cug="" a="" body="" call="" forward="" in="" is="" optional="" sent="" xml=""> <networkindicator> is mapped into t Network identity and the XML <cug> <cugl binary<="" closed="" code="" group="" iam="" interlock="" th="" user=""><th>dicator - CUG one IAM Closed one IAM Closed</th><th>call indicator ="11". The XML User Group interlock code</th></cugl></cug></networkindicator></cugcommunicationindicato>	dicator - CUG one IAM Closed one IAM Closed	call indicator ="11". The XML User Group interlock code
SIP Parameter	INVITE:	-	
values:	<cug></cug>		
	<networkindicator>[PIXIT]<cuginterlockbinarycode>[PIXIT]<cugcommunicationindicator>11</cugcommunicationindicator></cuginterlockbinarycode></networkindicator>	terlockBinaryC	
ISUP Parameter	IAM:		
values:	Optional Forward Call Indicator CUG call indicator = "11"		
	Closed User Group interlock code		la al-Dimanu O a da
	Binary code derived from INVITE XML b Network identity derived from INVITE X		
Comments:		UT	ISUP
Comments.	INVITE →	→	IAM
	180 Ringing ←	-	ACM
		ng tone	AOW
	200 OK INVITE ←	•	ANM
	ACK →	_	
		ersation	
	BYE		REL
	200 OK BYE →	<b>→</b>	RLC

TP516005	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	NOT PICS 5/7		
criteria:			
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</cug></cugcommunicationindicator></cug>		
	sent, the CUG call indicator is set to "00" no C		
SIP Parameter	and <cuginterlockbinarycode> into Closed Us INVITE:</cuginterlockbinarycode>	ser Group interlock code.	
values:	INVITE:   <cug></cug>		
	<pre><networkindicator>[PIXIT][PIXIT][PIXIT]10</networkindicator></pre>	erlockBinaryCode>	
ISUP Parameter	IAM:		
values:	Optional Forward Call Indicator CUG call indic		
	If optional forward call indicator have to be se		
Comments:	SIP SU		
	INVITE -	→ IAM	
	180 Ringing ←	<b>←</b> ACM	
	Ringin		
	200 OK INVITE ←	<b>←</b> ANM	
	ACK →		
	Conver		
	BYE •	← REL	
	200 OK BYE →	→ RLC	

TP516006	SIP reference: RFC 3261 [6]	ISUP reference:	
11 310000	on reference. At 6 5201 [6]	ES 283 027 [1], clause 7.4.1.1	
TSS reference:	SIP-ISUP/SS/CUG/	£ 4/	
SIP selection			
criteria:			
ISUP selection criteria:	NOT PICS 5/7		
	Manning of rough VMI alament in the receive	ad INIVITE augCommunicationIndicator value	
Test purpose:	Mapping of <cug> XML element in the receive "11". The PSTN/ISDN network does not supp</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 identity and the XML <cug> <cuginterlockbinarycode> is mapped into the IAM Closed User Group interlock code Binary code.</cuginterlockbinarycode></cug></networkindicator></cug></cug></cugcommunicationindicator>		
SIP Parameter	INVITE:		
values:	<cug></cug>		
	<networkindicator>[PIXIT]</networkindicator>		
	<pre><cuginterlockbinarycode>[PIXIT]</cuginterlockbinarycode></pre>		
	<pre><cugcommunicationindicator>11</cugcommunicationindicator></pre>		
ISUP Parameter			
values:			
Comments:	SIP SU	IT 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
		n IAM message whereby Calling Party Number
	parameter and the Generic Number are n	
	<ul> <li>Sends an INVITE message without</li> </ul>	ut the "P-Asserted-Identity header field", a
	"From header field" set to unavail	able@anonymous.invalid and without a Privacy
	Header field.	
SIP Parameter	INVITE: From <unavailable@anonymous:< th=""><th>•</th></unavailable@anonymous:<>	•
values:		
ISUP Parameter	IAM: no Calling party number	
values:	no Generic Number: "Additional ca	lling party number"
Comments:	ISUP/BICC SU	Γ SIP
	IAM →	→ INVITE
	ACM ←	← 180 Ringing
	Ringing tone	
	ANM ←	← 200 OK INVITE
		→ ACK
	Cor	nversation
	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	g party number received	
SIP Parameter values:	Ensure that when the SUT has received an IAM message whereby Calling Party Number parameter is <b>not</b> 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.  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 <b>←</b>	← 180 Ringing	
	Ringing tone		
	ANM ←	€ 200 OK INVITE	
	_	→ ACK	
		rsation	
	REL →	→ BYE	
	RLC <b>←</b>	← 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/	E3 203 027 [1], Clause 7.2.3.1.2.0	
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 <b>not</b> 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</unavailable@anonymous>		
ISUP Parameter	IAM: Generic Number: "additional calling party number", Nature of Address Indicator:		
values:	NoAS_VALUE APRI "presentation restricted"		
	ISUP/BICC SUT	SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	Ringing tone		
Comments:	ANM ←	← 200 OK INVITE	
		→ ACK	
		rsation	
	REL →	→ BYE	
	RLC <b>←</b>	← 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	ico di rograzii		
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 I contained whereby the Nature of Address Incept to "presentation allowed" and the Generic	dicator is set to NoAS_VALUE the APRI is	
	<ul> <li>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.</li> </ul>		
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	IAM: Calling party number APRI "presentati		
values:	no Generic Number: "Additional calling party number"		
Comments:	ISUP/BICC SUT	SIP	
	IAM →	→ INVITE	
	ACM <b>←</b>	← 180 Ringing	
	Ringing tone		
	ANM ←	← 200 OK INVITE	
	_	→ ACK	
		ersation	
	REL →	→ BYE	
	RLC <del>(</del>	← 200 OK BYE	

#### Table 40

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

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		
SIP Parameter	anonymous@anonymous.invalid and wi P-Asserted-Identity header field:	III Filvacy neader lield value ld .	
values:	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: anonymous@anonymous.invalid  Privacy header: "id".		
ISUP Parameter	IAM: Calling party number APRI "presentation	restricted"	
values:	no Generic Number: "Additional calling party number"		
Comments:	ISUP/BICC SUT	SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	Ringing tone		
	ANM ←	← 200 OK INVITE	
		→ ACK	
		ersation	
	REL →	→ BYE	
	RLC <b>←</b>	← 200 OK BYE	

TP601006	SIP referer	nce: RFC 3261 [6]	ES 28	ISUP reference: 3 027 [1], clause 7.2.3.1.2.6
TSS reference:	ISUP-SIP/SS/CLI/			2 2 1 [1], 0.0000 1 1 2 0 1 1 2 1 0
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			
	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 anonymous@ 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: anonymous@ anonymous.invalid  Privacy header: "id".			
ISUP Parameter		ty number APRI "presentat	tion restricte	nd by the network"
values:	no Generic Number: "Additional calling party number"			
Comments:	ISUP/BICC SUT SIP			
	IAM	<b>→</b>	→	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
		Ringing tone		
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
		_	<b>→</b>	ACK
			ersation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	<b>←</b>	+	200 OK BYE

### Table 41

	Values for test purpose TP601006			
	ISUP Parameter values: SIP Parameter values:			
VA_01	IAM	INVITE		
	NoAS_VALUE: "national (significant) number"(NDC+SN)	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	INVITE		
	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	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 I contained whereby the Nature of Address Incest to "presentation allowed" and the Generic	dicator is set to NoAS_VALUE the APRI is	
	"Tel or SIP URI" is set to PAIh_Addr "Tel or SIP URI" is set to FH_Addr_S or "id" is not included.	"P-Asserted-Identity header field", where the _SPEC_ID "From header field" where the SPEC_ID and without "Privacy Header field"	
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 values:	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	6 000 OK IN UTF	
	ANM ←	← 200 OK INVITE → ACK	
	Convo	rsation ACK	
	REL →	→ BYE	
	RLC +	€ 200 OK BYE	
	1/10	200 ON DIL	

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 received	and an additional calling party number	
	Ensure that when the SUT has received an Li contained whereby the Nature of Address Ind set to <b>presentation restricted</b> and the <b>Gene</b>		
	Sends an INVITE message with the		
		ne "addr-spec" is set to PAIh_Addr_SPEC_ID	
	"From header field" where the "addr-spec" is	s set to FH_Addr_SPEC_ID and with	
	"Privacy Header field =id".		
SIP Parameter	P-Asserted-Identity header field:		
values:	Tel or SIP URI: PAIh_Addr_SPEC_ID (Derived from Calling Party Number		
	parameter Address Signals)		
	From header field: addr-spec	ived from Conorio Number	
	Tel or SIP URI: FH_Addr_SPEC_ID (Deri parameter Address Signals (AcgPN))	ived from Generic Number	
	parameter Address Signals (Acgriv))		
	Privacy header: "id"		
ISUP Parameter	IAM: Calling party number APRI "presentation restricted"		
values:	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	<b>5</b>	
	ANM ←	← 200 OK INVITE	
		→ ACK	
		rsation	
	· · ==	→ BYE	
	RLC <del>(</del>	← 200 OK BYE	

Table 42

	Values for test purpose TP601007; TP601008			
Test purposes	ISUP Parameter values:	SIP Parameter values:		
VA_01	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)

TP602001	SIP reference:	RFC 3261 [6]		ISUP reference:
TCC vofevence:	ICUID CID/CC/UOLD/			[14], clause 7.4.10
TSS reference:	ISUP-SIP/SS/HOLD/			
SIP selection criteria:	PICS 8/4			
	DIOC 5/00			
ISUP selection criteria:	PICS 5/22			
	Each party can hold and	d ratriava tha ramata r	acetulia tha	antismad atata
Test purpose:	Each party can now and	a retrieve trie remote p	party in the t	commined 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	SDP: a=sendonly (put o			
values:		nitted (retrieve the call	)	
	o= <version inc<="" th=""><th>cremented&gt;</th><th></th><th></th></version>	cremented>		
ISUP Parameter				ROGRESS (put on hold)
values:				PROGRESS (retrieve the call)
Comments:	ISUP/BICC	MGC		SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
	ANM	<b>←</b>	+	200 OK INVITE
	CPG(hold)	<b>→</b>	<b>→</b>	INVITE(sendonly) 200 OK INVITE(recvonly)
	CPG(retrieve)	<b>→</b>	<b>→</b>	INVITE(sendrecv) 200 OK INVITE(sendrecv)
	CPG(hold)	<b>←</b>	<b>←</b> →	INVITE(sendonly) 200 OK INVITE(recvonly)
	CPG(retrieve)	+	<b>←</b> →	INVITE(sendrecv) 200 OK INVITE(sendrecv)

TP602002	SIP reference: RFC 3261 [6]	ISUP reference: [14], clause 7.4.10	
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	e other party	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= <version incremented=""></version>		
ISUP Parameter	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold)		
values:		ent indicator PROGRESS (retrieve the call)	
Comments:	ISUP/BICC MGC IAM → ACM ←	F SIP → INVITE ← 180 Ringing	
	CPG(hold) →	<ul><li>→ UPDATE(sendonly)</li><li>← 200 OK UPDATE(recevonly)</li></ul>	
	CPG(retrieve) →	<ul><li>→ UPDATE(sendrecv)</li><li>← 200 OK UPDATE(sendrecv)</li></ul>	

TP602003	SIP reference: RFC 3261 [6]	l		ISUP reference: [14], clause 7.4.10
TSS reference:	ISUP-SIP/SS/HOLD/			
SIP selection	PICS 8/4			
criteria:				
ISUP selection	PICS 5/22			
criteria:				
Test purpose:	HOLD indication in SDP in an UPDA	TE received	d	
	Ensure that a party can put the other the information necessary for process call previously put on hold.			
	The calling party should be able to pu	ut the other	party on I	hold
	The calling party should be able to re	etrieve the o	ther party	,
SIP Parameter	SDP: a=sendonly (put on hold)			
values:	a=sendrecv or omitted (retriev o= . <version incremented=""></version>	e the call)		
ISUP Parameter	ACM: called party status: no indication	on		
values:	CPG: Generic notification: remote ho Generic notification: remote reti			
Comments:	ISUP/BICC	MGCF		SIP
	IAM →		<b>→</b>	INVITE
	ACM ←		<b>←</b>	180 Ringing
	ANM ←		+	200 OK INVITE
	CPG(hold) ←		<del>(</del>	UPDATE(sendonly)
			<b>→</b>	200 OK UPDATE(recevonly)
	CPG(retrieve) ←		<b>←</b> →	UPDATE(sendrecv) 200 OK UPDATE(sendrecv)

TP602004	SIP reference: RFC 3261 [6]	ISUP reference:			
		[14], clause 7.4.10			
TSS reference:	ISUP-SIP/SS/HOLD/				
SIP selection criteria:	PICS 8/4 AND PICS 8/3				
ISUP selection criteria:	PICS 5/22				
Test purpose:	The SUT uses the UPDATE method to indicate	e HOLD in the SDP			
	Ensure that a party can put the other party on indicate the hold and retrieve state. Ensure th put on hold.				
	The calling party should be able to put the oth The calling party should be able to retrieve the				
SIP Parameter	UPDATE				
values:	SDP: a=sendonly (put on hold)				
	a=sendrecv or omitted (retrieve the call)				
	o= <version incremented=""></version>				
ISUP Parameter	CPG: Generic notification: remote hold Event				
values:		ent indicator PROGRESS (retrieve the call)			
Comments:	ISUP/BICC MGC	F SIP			
	IAM →	→ INVITE			
	ACM <b>←</b>	← 180 Ringing			
	ANM ←	← 200 OK INVITE			
	CPG(hold) →	→ UPDATE(sendonly)			
		← 200 OK UPDATE(recevonly)			
	CPG(retrieve) →	→ UPDATE(sendrecv)			
		<ul> <li>200 OK UPDATE(sendrecv)</li> </ul>			

TP602005	SIP reference:	RFC 3261 [6]		ISUP reference: [14], clause 7.4.10	
TSS reference:	ISUP-SIP/SS/HOLD/		l	E 4/	
SIP selection	PICS 8/4				
criteria:					
ISUP selection	PICS 5/22				
criteria:	<b>F</b>	-l (-l	and the the		
Test purpose:	Each party can hold an	a retrieve the remote p	arty in the d	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 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	,	
SIP Parameter	SDP: a=sendonly (put of				
values:		nitted (retrieve the call)	)		
	o= <version in<="" th=""><th></th><th></th><th></th></version>				
ISUP Parameter				ROGRESS (put on hold)	
values:				PROGRESS (retrieve the call)	
Comments:	ISUP/BICC	MGC	=	SIP	
	IAM	<b>→</b>	<b>→</b>	INVITE	
	ACM	<b>←</b>	<del>(</del>	180 Ringing	
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE	
	CPG(hold)	<b>→</b>	<b>→</b>	INVITE(sendonly)	
			<b>←</b>	200 OK INVITE(recvonly)	
	CPG(hold)	<b>←</b>	<b>←</b>	INVITE(inactive)	
			<b>→</b>	200 OK INVITE(inactive)	
	CPG(retrieve)	<b>→</b>	<b>→</b>	INVITE(recvonly)	
			+	200 OK INVITE(sendonly)	
	CPG(retrieve)	+	<b>←</b> →	INVITE(sendrecv) 200 OK INVITE(sendrecv)	

TP602006	SIP reference:	RFC 3261 [6]		ISUP reference: [14], clause 7.4.10	
TSS reference:	ISUP-SIP/SS/HOLD/		I	<b>L</b> 4/	
SIP selection	PICS 8/4				
criteria:					
ISUP selection	PICS 5/22				
criteria:	<u> </u>				
Test purpose:	Each party can hold ar	nd retrieve the remote p	arty in the d	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 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		d be able to retrieve the	e otner party	/	
values:	SDP: a=sendonly (put	on noid) mitted (retrieve the call)	<b>\</b>		
values.	o= <version in<="" th=""><th>`</th><th>)</th><th></th></version>	`	)		
ISUP Parameter			indicator PF	ROGRESS (put on hold)	
values:	Generic notification	on: remote retrieval eve	ent indicator	PROGRESS (retrieve the call)	
Comments:	ISUP/BICC	MGC	=	SIP	
	IAM	<b>→</b>	<b>→</b>	INVITE	
	ACM	<b>←</b>	<b>←</b>	180 Ringing	
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE	
	CPG(hold)	<b>→</b>	<b>→</b>	INVITE(sendonly)	
	, ,		<b>←</b>	200 OK INVITE(recvonly)	
	CPG(hold)	<b>←</b>	<b>←</b>	INVITE(inactive)	
			<b>→</b>	200 OK INVITE(inactive)	
	CPG(retrieve)	<b>←</b>	<b>←</b>	INVITE(recvonly)	
			<b>→</b>	200 OK INVITE(sendonly)	
	CPG(retrieve)	<b>→</b>	<b>→</b>	INVITE(sendrecv) 200 OK INVITE(sendrecv)	

# 6.3.2.3 Terminal portability (TP)

Void.

# 6.3.2.4 Conference calling (CONF)

TP604001	SIP reference: RFC 3261 [6]			N reference:
			ES 283 02	27 [1], clause 7.4.14
TSS reference:	ISUP-SIP/SS/CONF/			
SIP selection	PICS 8/2			
criteria:				
ISUP selection	PICS 5/10			
criteria:				
Test purpose:	Establish and disconnect a Conferen	се		
	Ensure that the SUT does not stop the streams if a CPG message Generic in Conference established Conference disconnected was received due to the CONF supp	notificatio	n indicator with the	
SIP Parameter				
values:				
ISUP Parameter	CPG: Generic notification = Confere	nce esta	blished	
values:	CPG: Generic notification = Confere	ence disc	onnected	
Comments:	ISUP/BICC		SUT	SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
	Ringing to	ne		
	ANM	←	<del>(</del>	200 OK INVITE
			<b>→</b>	ACK
		Conve	rsation	
	CPG(Conference established)	<b>→</b>		
	CPG(Conference disconnected)	→ Conve	rsation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	<b>←</b>	+	200 OK BYE

TP604002	SIP reference: RFC 3261 [6]	NGN reference: ES 283 027 [1], clause 7.4.14			
TSS reference:	ISUP-SIP/SS/CONF/	E3 203 027 [1], Clause 7.4.14			
SIP selection	PICS 8/2				
criteria:	1 100 0/2				
ISUP selection	PICS 5/10				
criteria:					
Test purpose:	Isolate and reattach a Conference				
	Ensure that the SUT stop the temporarily CPG message Generic notification indica Isolated reattached was received due to the CONF supplement				
SIP Parameter	SDP: a= sendonly	The state of the s			
values:	SDP: a= sendrecv				
ISUP Parameter	CPG: Generic notification = Conference	established			
values:	CPG: Generic notification = Isolated				
	CPG: Generic notification = Reattached				
Comments:	CPG: Generic notification = Conference ISUP/BICC				
Comments:	IAM →	SUT SIP → INVITE			
	ACM +	◆ 180 Ringing			
	Ringing tone	180 Kinging			
	ANM •	← 200 OK INVITE			
	7 (I VIVI	→ ACK			
	C	onversation			
	CPG(Conference established) →	onvoidation			
	,				
	CPG(Isolated) →	→ INVITE(sendonly)			
		<ul> <li>200 OK INVITE(recvonly)</li> </ul>			
		→ ACK			
	CPG(Reattached) →	→ INVITE(sendrecv)			
		← 200 OK INVITE(sendrecv) → ACK			
		7 AUN			
	CPG(Conference disconnected) →				
	_	onversation			
	REL →	→ BYE			
	RLC ←	← 200 OK BYE			

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:	NOT BIOG 5/40		
ISUP selection criteria:	NOT PICS 5/10		
Test purpose:	Mapping of isolated and reattached	d not suppo	orted
	Ensure that the MGCF can receive "reattached", no mapping on the S  No mapping, no disrupting the	P side and	·
SIP Parameter values:	No mapping		
ISUP Parameter	CPG: Generic notification = Confe	rence esta	blished
values:	CPG: Generic notification = isolate		
	CPG: Generic notification = reatta	ched	
	CPG: Generic notification = Confe		
Comments:	ISUP/BICC	-	SUT SIP
	IAM	<b>→</b>	→ INVITE
	ACM	<b>←</b>	← 180 Ringing
	Ringing ton		<b>4</b> 000 OK IN UTE
	ANM	<b>←</b>	€ 200 OK INVITE
		0	→ ACK
	CDC/Conformed autoblished		ersation
	CPG(Conference established)	<b>→</b>	
	CPG(Isolated)	<b>→</b>	
	CPG(Reattached)	<b>→</b>	
	CPG(Conference disconnected)	→ Conve	ersation
	REL RLC	→ ←	→ BYE ← 200 OK BYE
L	1.1.20	_	1 200 010 012

TP604004	SIP reference: RFC 3261	[6]		ISUP reference:
		. •		ec. Q.1912.5 [32], annex B.14 Rec. Q.734.1 [34], clause 1.7
TSS reference:	ISUP-SIP/SS/CONF/			<u> </u>
SIP selection				
criteria:				
ISUP selection criteria:	NOT PICS 5/10			
Test purpose:	No mapping of generic notifications	s no chana	a tha sassia	n state
rest purpose.	Two mapping or generic notifications	s no chang	e ine sessio	II state
	Ensure that the MGCF can receive	in a CPG	he Generic	notifications is "other party
	added" or "other party isolated" or			
	conference floating" or " other party			
	side and the call is not disrupted.			
		010		
CID Dozomotor	No mapping, no disrupting the	SIP proced	ure.	
SIP Parameter values:	No mapping			
ISUP Parameter	CPG: Generic notification = Confe	rence esta	blished	
values:	CPG: Generic notification = other			
	CPG: Generic notification = other	party isolat	ed	
	CPG: Generic notification = other		ached	
	CPG: Generic notification = other		_	
	CPG: Generic notification = other			
	CPG: Generic notification = Confe CPG: Generic notification = Confe			
Comments:	ISUP/BICC		SUT	SIP
Comments.	IAM	<b>→</b> `	- <del>-</del>	-
	ACM	É	·	
	Ringing ton	=	_	100111191119
	ANM	<b>←</b>	•	200 OK INVITE
			-	→ ACK
		Conve	rsation	
	CPG(Conference established)	<b>→</b>		
	CPG(other party added)	<b>→</b>		
	Ci G(other party added)			
	CPG(other party isolated)	<b>→</b>		
	CPG(other party reattached)	<b>→</b>		
	CPG(other party split)	<b>→</b>		
	CPG(other party disconnected)	<b>→</b>		
	CPG(Conference floating)	<b>→</b>		
	CPG(Conference disconnected)	→ Conve	rsation	
	REL	→ CONVE	13aliUH -	▶ BYE
	RLC	É	4	- 200 OK BYE
	I NEO			200 OK DIL

TP604005	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.4.1.1.1		
TSS reference:	ISUP-SIP/SS/CONF/	20 200 027 [1], 010000 71-711111		
SIP selection	PICS 1/1			
criteria:	. 100 1/1			
ISUP selection				
criteria:				
Test purpose:	Conference notification information is mapped	l into "conference established"		
	Upon the receipt of a conference information document with the <conference-state-type></conference-state-type>			
	element active is set to "true", the MGCF shall	I send a CPG message to the CS side with a		
010.0	notification "conference established".			
SIP Parameter		bscription-State contains active;		
values:	expires=xxxx	1.		
	application/conference-info+xml	l.		
	entity=conference URI st	tate="full" version="x"		
	<conference-state></conference-state>	tato- raii voroion- x		
	<user-count>2<th>-count&gt; if present</th></user-count>	-count> if present		
	<active>true</active>			
	<users></users>	·		
	<user entity="ISUPx" th="" u<=""><th></th></user>			
		endpoint ISUPx URI		
	<status>connected</status>			
	<joining-method>dialed-out<!-- joining-method--> <media <="" id="1" th=""></media></joining-method>			
	<media id="1&lt;br"><status>sendrecv</status></media>			
	<user <="" entity="SIPx" state="full" th="" uri=""></user>			
	<endpoint entity="endpoint" sipx="" th="" uri<=""></endpoint>			
	<pre><status>connected</status></pre>			
	<pre><joining-method>dialed-in</joining-method></pre> /joining-method>			
	<media <="" id="1" th=""></media>			
		endrecv		
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:	
TCC reference.	IOUR OID/OC/CONE/	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	d into "conforance actablished"	
SIP Parameter values:	Upon the receipt of a conference information of element status of endpoint-status-type is set to before and the Contact URI in the element en PSTN/ISDN participant, the MGCF shall send notification "other party added".  NOTIFY 1: See test case TP604006	document with the <endpoint-type> and the order of the connected and it was not set to "on-hotity is not the address of the served</endpoint-type>	
	application/conference-info+xm <conference-info> entity=conference URI s <conference-state> <user-count>y <user <endpoint="" entity="&lt;status" uf="">conr <joining-meth <="" <media="" id="1" th=""><th>tate="full" version="x"  -count&gt; if present  RI state="full" -endpoint SIPx URI nected lod&gt;dialed-out</th><th></th></joining-meth></user></user-count></conference-state></conference-info>	tate="full" version="x"  -count> if present  RI state="full" -endpoint SIPx URI nected lod>dialed-out	
ISUP Parameter	CPG(other party added)		
values:	loup	M005	
Comments:	ISUP IAM  ACM  ←  ACM  ←	MGCF SIP  → INVITE  ← 180 Ringing  ← 200 OK INVITE	
	CPG(Conference established) ←	<ul><li>NOTIFY 1</li><li>→ 200 OK NOTIFY</li></ul>	
	CPG(other party added) ←	<ul><li>NOTIFY 2</li><li>→ 200 OK NOTIFY</li></ul>	
	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	PICS 1/1	
criteria:		
ISUP selection		
criteria:		
Test purpose:	Conference notification information is mapped	d into "isolated"
	Upon the receipt of a conference information of element status of endpoint-status-type is set to before and the Contact URI in the element en participant, the MGCF shall send a CPG mess "isolated".	to "on-hold" and it was set to "connected" tity is the address of the served PSTN/ISDN
SIP Parameter values:	application/conference-info+xm <conference-info> entity=conference URI s <conference-state> <user-count>2 <user <endpoint="" entity="&lt;status" t="">on-h <joining-meth <="" <media="" id="1" th=""><th>tate="full" version="x" r-count&gt; if present  URI state="full" endpoint ISUPx URI old od&gt;dialed-out</th></joining-meth></user></user-count></conference-state></conference-info>	tate="full" version="x" r-count> if present  URI state="full" endpoint ISUPx URI old od>dialed-out
ISUP Parameter	CPG(isolated)	
values:		
Comments:	ISUP IAM  ACM  ←  ACM  ←	MGCF SIP  → INVITE  ← 180 Ringing  ← 200 OK INVITE
	CPG(Conference established)	<ul><li>NOTIFY 1</li><li>→ 200 OK NOTIFY</li></ul>
	CPG(isolated) ←	<ul><li>NOTIFY 2</li><li>→ 200 OK NOTIFY</li></ul>
	REL → RLC ←	→ BYE ← 200 OK BYE

TSS reference:  SUP-SIP/SS/CONF/    SIP selection criteria:   PICS 1/1	TP604008	SIP reference: RFC 3261 [6]	ISUP reference:
SIP selection criteria:   ISUP selection criteria:   Conference notification information is mapped into "other party 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 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".    SIP Parameter values:</endpoint-type>			ES 283 027 [1], clause 7.4.1.1.1
Criteria:   ISUP selection criteria:   ISUP selection criteria:   Conference notification information is mapped into "other party isolated"   Upon the receipt of a conference information document with the cendpoint-type> and the element status of and point-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".    SIP Parameter values:			
Test purpose:  Conference notification information is mapped into "other party 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 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".  SIP Parameter values:  NOTIFY 1: See test case TP604006 NOTIFY 2: Event contains conference; Subscription-State contains active application/conference-info+xml:</endpoint-type>	criteria:	PICS 1/1	
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".  SIP Parameter values:  NOTIFY 1: See test case TP604006  NOTIFY 2: Event contains conference; Subscription-State contains active application/conference-info+xml:</endpoint-type>			
values:       NOTIFY 2:       Event contains conference; Subscription-State contains active application/conference-info+xml:		Upon the receipt of a conference information element status of endpoint-status-type is set to before and the Contact URI in the element en PSTN/ISDN participant, the MGCF shall send notification "other party isolated".	document with the <endpoint-type> and the to "on-hold" and it was set to "connected" tity is not the address of the served</endpoint-type>
CPG(other party isolated)   CPG(other party isolated)   COMMENTS:		NOTIFY 2: Event contains <b>conference</b> ; Su application/conference-info+xm < conference-info> entity=conference URI s < conference-state> < user-count>2 <th>al:  state="full" version="x"  r-count&gt; if present  RI state="full" endpoint SIPx URI sold sod&gt;dialed-out "</th>	al:  state="full" version="x"  r-count> if present  RI state="full" endpoint SIPx URI sold sod>dialed-out "
Comments:  ISUP IAM ACM ACM ACM ACM CPG(Conference established)  CPG(other party isolated)  MGCF SIP INVITE 180 Ringing 200 OK INVITE  NOTIFY 1 200 OK NOTIFY  NOTIFY 2 200 OK NOTIFY			onaroov volutade
RLC ← 200 OK BYE		IAM ACM ACM ← CPG(Conference established) ← CPG(other party isolated) ← REL	<ul> <li>→ INVITE</li> <li>← 180 Ringing</li> <li>← 200 OK INVITE</li> <li>← NOTIFY 1</li> <li>→ 200 OK NOTIFY</li> <li>← NOTIFY 2</li> <li>→ 200 OK NOTIFY</li> <li>→ BYE</li> </ul>

TP604009	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.4.1.1.1	
TSS reference:	ISUP-SIP/SS/CONF/	20 200 02. [.], 0.0000	
SIP selection	PICS 1/1		
criteria: 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".</endpoint-type>		
SIP Parameter	NOTIFY 1: see test case TP604006		
values:	NOTIFY 2: Event contains <b>conference</b> ; Subscription-State contains <b>active</b>		
	application/conference-info+xml:		
	entity=conference URI state="full" version="x"		
	<conference-state></conference-state>		
	<user-count>2</user-count> if present		
	<users></users>	1151	
	<user <endpoint="" entity="endpoint" isupx="" state="full" th="" uri="" uri<=""></user>		
	<pre><status>on-hold</status></pre>		
	<pre><joining-method>dialed-out</joining-method></pre> /joining-method>		
	<media <="" id="1" th=""></media>		
	<status>sendrecv</status>		
	NOTIFY 3: Event contains <b>conference</b> ; Subscription-State contains <b>active</b>		
	application/conference-info+xml: <conference-info></conference-info>		
	entity=conference URI state="full" version="x"		
	<conference-state></conference-state>		
	<user-count>2<th>r-count&gt; if present</th></user-count>	r-count> if present	
	<user's></user's>	LIDI state "full"	
		entity=ISUPx URI state="full" ndpoint entity=endpoint ISUPx URI	
	<status>connected</status> <joining-method>dialed-out</joining-method>		

TP604010	SIP reference: RFC 3261 [6]		SUP reference: 27 [1], clause 7.4.1.1.1
TSS reference:	ISUP-SIP/SS/CONF/		£ 4/
SIP selection	PICS 1/1		
criteria:			
ISUP selection			
criteria:			
Test purpose:	Conference notification information is mappe		
	Upon the receipt of a conference information		
	element status of endpoint-status-type is set "on-hold" before and the Contact URI in the e		
	PSTN/ISDN participant, the MGCF shall send		
	notification "other party reattached".		
SIP Parameter	NOTIFY 1: See test case TP604006		
values:	NOTIFY 2: Event contains conference; Se	ubscription-State	contains active
	application/conference-info+xn	nl:	
	<conference-info></conference-info>		
	entity=conference URI	state="full" versio	on="x"
	<conference-state></conference-state>	r counts if proce	nt
	<user-count>2<th>i-count&gt; ii prese</th><th>TIL.</th></user-count>	i-count> ii prese	TIL.
	<user entity="SIPx" th="" u<=""><th>RI state="full"</th><th></th></user>	RI state="full"	
	<endpoint entity:<="" th=""><th></th><th>JRI</th></endpoint>		JRI
	<status>on-l</status>	nold	
			joining-method>
	<media <="" full"="" id="1&lt;/th&gt;&lt;th&gt;&lt;/th&gt;&lt;th&gt;&lt;/th&gt;&lt;/tr&gt;&lt;tr&gt;&lt;th&gt;&lt;/th&gt;&lt;th&gt;NOTIFY 3: Event contains &lt;b&gt;conference&lt;/b&gt;; Se&lt;/th&gt;&lt;th&gt;sendrecv&lt;/status&lt;/th&gt;&lt;th&gt;&lt;/th&gt;&lt;/tr&gt;&lt;tr&gt;&lt;th&gt;&lt;/th&gt;&lt;th&gt;application/conference-info+xn&lt;/th&gt;&lt;th&gt;&lt;/th&gt;&lt;th&gt;Contains active&lt;/th&gt;&lt;/tr&gt;&lt;tr&gt;&lt;th&gt;&lt;/th&gt;&lt;th colspan=3&gt;&lt;pre&gt;application/conference-info+xffil. &lt;/pre&gt;&lt;/th&gt;&lt;/tr&gt;&lt;tr&gt;&lt;th&gt;&lt;/th&gt;&lt;th colspan=3&gt;entity=conference URI state=" th="" version="x"></media>		
	<conference-state></conference-state>		
	<user-count>2</user-count> if present		
	<user <="" lidi="" ontity="SIDv" state="full" th=""></user>		
	<user <endpoint="" entity="endpoint" sipx="" state="full" th="" uri="" uri<=""></user>		
	<pre><endpoint <="" entity="endpoint" ort="" pre="" sipx=""> <pre><status>connected</status></pre></endpoint></pre>		
			joining-method>
	<media <="" id="1&lt;/th&gt;&lt;th&gt;" th=""><th></th></media>		
		sendrecv <th>&gt;</th>	>
ISUP Parameter	CPG(other party reattached)		
values: Comments:	ISUP	MGCF	SIP
Comments.	IAM →	₩GCF	INVITE
	ACM +	÷	180 Ringing
	ACM ←	<del>-</del>	200 OK INVITE
	-		
	CPG(Conference established) ←	<b>←</b>	NOTIFY 1
		<b>→</b>	200 OK NOTIFY
	CPG(other party isolated)	<b>←</b>	NOTIFY 2
		<b>→</b>	200 OK NOTIFY
	ODO(athan mark	-	NOTIFY
	CPG(other party reattached)	<b>←</b>	NOTIFY 3
		7	200 OK NOTIFY
	REL →	<b>→</b>	BYE
	RLC +	÷	200 OK BYE
1	1	<u>*</u>	

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	PICS 1/1			
criteria:				
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".</endpoint-type>			
SIP Parameter	NOTIFY 1: See test case TP604006	and the Chata and the satisfactors		
values:	NOTIFY 2: Event contains conference; Subscription-State contains active application/conference-info+xml:			
	<pre><conference-state> <user-count>y</user-count></conference-state></pre>			
	<users></users>	•		
	<status>conne <joining-methe <media <="" id="1" th=""><th>endpoint SIPx URI ected</th></media></joining-methe </status> od>dialed-out	endpoint SIPx URI ected		
	NOTIFY 3: Event contains <b>conference</b> ; Su application/conference-info+xml	bscription-State contains active		
	<pre><conference-info>     entity=conference URI st     <conference-state></conference-state></conference-info></pre>	tate="full" version="x"		
	<users></users>			
	<user <endpoint="" <status="" entity="endpoint" sipx="" state="full" uri="">disconnected <joining-method>dialed-out</joining-method> <disconnection-method>booted<disconnection-method></disconnection-method> <media <status="" id="1">sendrecv</media></disconnection-method></user>			
ISUP Parameter	CPG(other party disconnected)			
values:				
Comments:		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) ←	<ul><li>NOTIFY 3</li><li>→ 200 OK NOTIFY</li></ul>		
	REL →	→ BYE		
	RLC <del>C</del>	€ 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 map	pped to PSTN	
	Upon the receipt of a conference information of	document the conference notification	
SIP Parameter values:	information is not mapped to the PSTN side. No NOTIFY is sent to the ISDN user.  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:		
ISUP Parameter values:	CPG(other party added)	endrecv	
Comments:	ISUP IAM → ACM ← ACM ←	MGCF SIP  → INVITE  ← 180 Ringing  ← 200 OK INVITE   ← NOTIFY 1  → 200 OK NOTIFY  ← NOTIFY 2  → 200 OK NOTIFY	
	REL → RLC ←	→ BYE ← 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 wi	hen a 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 co	onference URI	
	Refer-To contains the URI of I	ISUPx, method=invite	
	Referred-By contains SIP or to	el URI of <b>SIPx</b>	
ISUP Parameter	CPG(Conference established)		
values:			
Comments:	ISUP	MGCF SIP	
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	ACM ←	← 200 OK INVITE	
		<b>←</b> REFER	
		→ 403 Forbidden	

# 6.3.2.5 Three Party service (3PTY)

TP605001	SIP reference: RFC 3261 [6	j]	NGN reference:	
			ES 283 027 [1], clause 7.4.15	
TSS reference:	ISUP-SIP/SS/3PTY/			
SIP selection	PICS 8/2			
criteria:				
ISUP selection	PICS 5/5 AND PICS 5/18			
criteria:				
Test purpose:	The media stream is resumed if a 31	PTY is esta	ablished	
	E # ## OUT #	P		
			m put on hold while the GPG (hold) was	
			a-line in the SDP is set to "sendrecv" if a	
OID Danamatan	CPG (Conference established) was	receivea		
SIP Parameter	SDP: a= sendonly			
values:	SDP: a= sendrecv			
ISUP Parameter	CPG: Generic notification = remote			
values:	CPG: Generic notification = Confer			
	CPG: Generic notification = Confer			
Comments:	ISUP/BICC	_	SUT SIP	
		<b>→</b>	→ INVITE	
	710111	<b>←</b>	← 180 Ringing	
	Ringing tone			
	ANM	<b>←</b>	← 200 OK INVITE	
			→ ACK	
		Conve	rsation	
	CPG(hold)	<b>→</b>	→ INVITE(sendonly)	
			← 200 OK INVITE(recvonly)	
			→ ACK	
	CPG(Conference established)	<b>→</b>	→ INVITE(sendrecv)	
		-	€ 200 OK INVITE(sendrecv)	
			→ ACK	
			AUN	
	CPG(Conference disconnected)	<b>→</b>		
	CFG(Contenence disconnected)	Conve	rootion	
	REL	Onvei		
		<del>7</del> ←		
	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	PICS 8/1			
criteria:				
ISUP selection	PICS 5/5 AND PICS 5/18			
criteria:				
Test purpose:	Establish and disconnect a 3PTY s	ession. SL	DP conveyed in an UPDATE request:	
	Ensure that the SUT resumes the media stream put on hold while the GPG (hold) was received and sends an UPDATE containing an a-line in the SDP is set to "sendrecv" if a CPG (Conference established) was received			
SIP Parameter	SDP: a= sendonly			
values:	SDP: a= sendrecv			
ISUP Parameter	CPG: Generic notification = remot	e hold		
values:	CPG: Generic notification = Confe	rence esta	ablished	
	CPG: Generic notification = Confe	rence disc	connected	
Comments:	ISUP/BICC		SUT SIP	
	IAM	<b>→</b>	→ INVITE	
	ACM	<b>←</b>	← 180 Ringing	
	Ringing ton	е		
	ANM	<b>←</b>	← 200 OK INVITE → ACK	
	CPG(hold)	Conve	ersation  → UPDATE(sendonly)  ← 200 OK UPDATE(recvonly)	
	CPG(Conference established)	<b>→</b>	<ul><li>→ UPDATE(sendrecv)</li><li>← 200 OK UPDATE(sendrecv)</li></ul>	
	CPG(Conference disconnected)	→ Conve	ersation	
	REL	<b>→</b>	→ BYE	
	RLC	<b>←</b>	← 200 OK BYE	

TP605003	SIP reference: RFC 3261 [6	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 establis	hed" and "C	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	CPG: Generic notification = remote	hold	
values:	CPG: Generic notification = Confer CPG: Generic notification = Confer		
Comments:	ISUP/BICC	SU	
	IAM	<b>→</b>	→ INVITE
	ACM	<b>←</b>	← 180 Ringing
	Ringing tone	9	
	ANM	<b>←</b>	← 200 OK INVITE
		•	→ ACK
	CPG(hold)	Convers	ation
	CPG(Conference established)	<b>→</b>	
	CPG(Conference disconnected)	→ Convers	ation
	REL	<b>→</b>	→ BYE
	RLC	<del>-</del>	← 200 OK BYE

## 6.3.2.6 Connected line identification (COL)

TP606001	SIP reference: RF	C 3261 [6]	ES 28	33 027 [1], clause 7.4.2
TSS reference:	ISUP-SIP/SS/COL /			
SIP selection	NOT PICS 5/3			
criteria:				
ISUP selection				
criteria:				
Test purpose:	IAM with OFCI "connected	l line request" rece	ived, no mapping	7
			•	ward call indicator, connected
	line requested, continue w	ithout disrupting th	e SIP or ISUP si	gnalling procedure.
SIP Parameter	No mapping			
values:				
ISUP Parameter	IAM: Optional Forward cal	I indicator "Connec	ted line request"	
values:				
Comments:				
	ISUP		SUT	SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	+	180 Ringing
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Con	versation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	<b>←</b>	<b>←</b>	200 OK BYE

TP606002	SIP reference: RFC 3261	[6]	ES 28	3 027 [1], clause 7.4.2
TSS reference:	ISUP-SIP/SS/COL /			
SIP selection	PICS 5/3			
criteria:				
ISUP selection				
criteria:				
Test purpose:	IAM with oFCi "connected line req	uest" received, INVI	TE is se	nt contains the "from-change"
	tag in the Supported header			
	Ensure that the SUT if the IAM is a			
	line requested, the "from-change"	tag is included in the	e Suppoi	rted header in the sent INVITE.
SIP Parameter	INVITE: Supported: from-change			
values:				
ISUP Parameter	IAM: Optional Forward call indicate	or "Connected line r	equest"	
values:				
	ISUP	SUT		SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conversation	1	
	REL	<b>→</b>	<b>→</b>	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	NOT PICS 1/5		
criteria:			
Test purpose:	The P-Asserted-Identity is mapped into the connected number "national (significant) number"		
	NDC+ SN has been received and	ning a URI with an identity in the format "+" CC+ Privacy header field was received and the priv-	
	value is set to "none"	Filivacy fleader field was received and the priv-	
	in the ANM or CON is included the <b>Connecte</b> If CC encoded in the URI is <b>equal</b> to the CC onext BICC/ISUP node is located in the same of	of the country where MGCF is located AND the	
	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: <b>NDC+SN</b>		
	Generic number parameter not present		
SIP Parameter values:	1XX or 2XX response: P-Asserted-Identity her	ader field Tel URL containing an URI in the	
ISUP Parameter	IAM: Optional Forward Call Indicators, Conne	cted Line Identity Request indicator" =	
values:	"requested"	,	
	ANM;		
	Connected number parameter		
	Address presentation restricted parameter	er = '00'B	
	Nature of address indicator = '0000011'B		
	Numbering plan indicator = '001'B		
	Screening indicator = '11'B		
	Address signals = PIXIT	OOF OID	
	ISUP M	GCF SIP	
	IAW	→ INVITE ← SIP_MESSAGE_VA	
	CASE A	SII _WESSAGE_VA	
	ACM ←		
	ANM		
	CASE B		
	CON +		
		ersation	
	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	PICS 1/5			
criteria:				
Test purpose:	The P-Asserted-Identity is mapped into the connected number "international number"			
	been requested by the calling party by parsin the "Connected Line Identity Request indicate 2XX message defined as SIP_MESSAGE_V	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		
	NDC+ SN has been received and	ining a URI with an identity in the format "+" CC+ Privacy header field was received and the priv-		
	in the ANM or CON is included the <b>Connected number Parameter</b> .  If CC encoded in the URI is <b>not equal</b> 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 = <b>Presentation allowed</b> Nature of address indicator = <b>International number</b> 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	Generic number parameter not present  1XX or 2XX response: P-Asserted-Identity header field Tel URL containing an URI in the			
values:	format "+"CC+NDC+SN			
ISUP Parameter	IAM: Optional Forward Call Indicators, Conne	ected Line Identity Request indicator" =		
values:	"requested"	, ,		
	ANM;			
	Connected number parameter			
	Address presentation restricted parameter			
	Nature of address indicator = '0000011'E	3		
	Numbering plan indicator = '001'B			
	Screening indicator = '11'B			
	Address signals = PIXIT	ACCE CID		
	ISUP IAM +	MGCF SIP → INVITE		
	<del>7</del>	SIP_MESSAGE_VA		
	CASE A	5E00/(0L_v//		
	ACM ←			
	ANM ←			
	CASE B			
	CON ←			
		versation		
	REL -	→ BYE		
	RLC ←	← 200 OK BYE		

TP606005	SIP reference: RFC 3261 [6]		ISUP reference:	
		ES 28	3 027 [1], clause 7.4.2	
TSS reference:	ISUP-SIP/SS/COL /			
SIP selection	PICS 5/3			
criteria:				
ISUP selection	PICS 1/5			
criteria:				
Test purpose:	P-Asserted-Identity not received, a connected	number "addres	ss not available" is sent	
	Ensure that the SUT in Idle state, on receipt o been requested by the calling party by parsing the "Connected Line Identity Request indicato 2XX message defined as SIP_MESSAGE_VA	the "Optional For r" is set to "reque	orward Call Indicators" field and	
	no P-Asserted-Identity header field			
	In the ANM or CON is included the <b>Connecte</b>	d number Parar	meter.	
	Address presentation restricted parameter Screening indicator = Network Provided Address signals omitted	= Address not	available	
	Generic number parameter not present			
SIP Parameter	1XX or 2XX response: P-Asserted-Identity hea	ader field is not r	present	
values:	That or End the period to the transfer of	, a.cc		
ISUP Parameter	IAM: Optional Forward Call Indicators, Connected Line Identity Request indicator" =			
values:	"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			
	ISUP M	GCF	SIP	
	IAM →	<b>→</b>	INVITE	
		+	SIP_MESSAGE_VA	
	CASE A			
	ACM ←			
	ANM ←			
	CASE B			
	CON			
	Conv	ersation		
	REL →	<b>→</b>	BYE	
	RLC ←	+	200 OK BYE	

Values for tests purposes TP606003 to TP606005				
VA_01	180 Ringing			
VA_02	183 Session progress			
VA_03	200 OK			

TP606006	SIP reference: RFC 3261 [6]	ISUP reference:			
TOO refere		ES 283 027 [1], clauses 7.4.2 and 7.5.2			
TSS reference: SIP selection	ISUP-SIP/SS/COL / PICS 5/3				
criteria:	1 100 0/0				
ISUP selection					
criteria:	Intermedian of Francis I is a USB ATE	An additional appropriate to the state of th			
Test purpose:	Interworking of From header in the UPDATE. ANM or CON	An additional connected number is sent in the			
	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 an the "Connected Line Identity Request indicator" is set to "requested", on receipt of a 1XX or 2XX message with				
	the option tag "from-change" is received a Privacy header field was received and t	ne priv-value is set to PRIV_VALUE			
	An UPDATE request is received containing a with an identity in the format "+" CC+ NDC+ s	SN then			
	the ANM or CON	the UPDATE request to the Generic number in			
	In the ANM or CON is included the <b>Connecte</b> Numbering plan indicator = ISDN/Telepho Address presentation restricted paramete Screening indicator = Network Provided Address signals derived from the P-Asser	ny numbering plan r = Presentation restricted			
	In the ANM or CON is included the <b>Generic</b> I	•			
	Number Qualifier = additional connected in				
	Address presentation restricted paramete				
		Numbering plan indicator = ISDN numbering plan Screening indicator = user provided, not verified			
OID D	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	04/D			
	Address presentation restricted parameter = Nature of address indicator = '0000011'B	OLR			
	Numbering plan indicator = '000'B				
	Screening indicator = '11'B				
	Address signals = derived from the From hea				
	ISUP NAM →	IGCF SIP → INVITE			
	IZIVI <b>7</b>	→ IINVII E			
	CASE A				
	ACM <b>←</b>	← 180 Ringing			
		← 200 OK INVITE			
		→ ACK			
	ANM ←	<ul><li>← UPDATE</li><li>→ 200 OK UPDATE</li></ul>			
	CASE B				
	ACM ←	← 183 Session Progress			
	_	€ 200 OK INVITE			
		→ ACK			
	_	← UPDATE			
	ANM ←	→ 200 OK UPDATE			
	]				

TP606006	SIP reference: RFC 3261 [6]			ISUP reference:
			ES 283 027	[1], clauses 7.4.2 and 7.5.2
	CASE C			
			<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
			<b>←</b>	UPDATE
	CON	<b>-</b>	<b>→</b>	200 OK UPDATE
		Conve	ersation	
	REL -	<b>→</b>	<b>→</b>	BYE
	RLC	<b>F</b>	<b>←</b>	200 OK BYE

TP606007	SIP reference: RFC 3261 [6]	ISUP reference:			
TSS reference:	ISUP-SIP/SS/COL /	ES 283 027 [1], clauses 7.4.2 and 7.5.2			
SIP selection	PICS 5/22				
criteria:					
ISUP selection					
criteria: Test purpose:	Interworking of P-Asserted-Identity and From	header. The Connected number and the			
rest purpose:	additional connected number is presentation a				
	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 with				
	the P-Asserted-Identity header field contai NDC+ SN and the option tag "from-change a Privacy header field is not included	ining a URI with an identity in the format "+" CC+ e" is received			
	An UPDATE request is received containing a with an identity in the format "+" CC+ NDC+ S - map the From header field received in the ANM				
	In the ANM or CON is included the <b>Connecte</b> Numbering plan indicator = ISDN/Telepho Address presentation restricted parameter Screening indicator = Network Provided Address signals derived from the P-Assert	ny numbering plan r = Presentation allowed			
	In the ANM or CON is included the <b>Generic number parameter</b> 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	INVITE: Supported: from-change				
values:	1XX or 2XX response: P-Asserted-Identity her format "+"CC+NDC+SN UPDATE: From header, P-Asserted-Identity	ader field Tel URL containing an URI in the			
ISUP Parameter	IAM: Optional Forward Call Indicators, Conne	cted Line Identity Request indicator" =			
values:	"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 = " Numbering plan indicator = '001'B	'00"B			
	Screening indicator = '11'B				
	Address signals = derived from the From head				
		IGCF SIP			
	IAM →	→ INVITE			
	CASE A				
	ACM <b>←</b>	<ul><li>← 180 Ringing</li><li>← 200 OK INVITE</li><li>→ ACK</li></ul>			
		← UPDATE			
	ANM ←	→ 200 OK UPDATE			
	CASE B				
	ACM ←	← 183 Session Progress			

TP606007	SIP reference: RFC 32	261 [6]	ES 283	ISUP reference: 027 [1], clauses 7.4.2 and 7.5	5.2
		_	-	<ul><li>← 200 OK INVITE</li><li>→ ACK</li><li>← UPDATE</li></ul>	
	ANM	<b>←</b>	-	→ 200 OK UPDATE	
	CASE C		-	<ul><li>← 200 OK INVITE</li><li>→ ACK</li><li>← UPDATE</li></ul>	
	CON	<b>←</b>		→ 200 OK UPDATE	
	REL RLC	<b>→</b>		<ul><li>→ BYE</li><li>← 200 OK BYE</li></ul>	

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	PICS 5/22				
criteria:					
ISUP selection					
criteria:					
Test purpose:	Interworking of P-Asserted-Identity to the connected number if no UPDATE was received				
	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 with				
	the P-Asserted-Identity header field contain NDC+ SN and the option tag "from-change a Privacy header field is not included	ning a URI with an identity in the format "+" CC+ " is received and			
	When the 200 OK was received, start timer $T_T$	IR1			
	An UPDATE request is not received				
	After T <sub>TIR1</sub> was expired the ANM or CON is sent				
	In the ANM or CON is included the <b>Connected number Parameter</b> .				
	Numbering plan indicator = ISDN/Telephor				
	Address presentation restricted parameter	= Presentation allowed			
	Screening indicator = Network Provided				
	Address signals derived from the P-Asserted-I	dentity in the 200 OK INVITE. A Generic			
SIP Parameter	Number parameter is not present INVITE: Supported: from-change				
values:	1XX or 2XX response: P-Asserted-Identity hea	ader field Tel URL containing an URL in the			
	format "+"CC+NDC+SN				
ISUP Parameter	IAM: Optional Forward Call Indicators, Connec	cted Line Identity Request indicator" =			
values:	"requested"				
	ANM or CON				
	Connected number parameter	00110			
	Address presentation restricted parameter = "(	nnR			
	Numbering plan indicator = '001'B Screening indicator = '11'B				
	Address signals = PIXIT				
<u> </u>	Audiess signals - LIALL				

TP606008	SIP reference	e: RFC 3261 [6]	F0 000		SUP reference:
				027	[1], clauses 7.4.2 and 7.5.2
	ISUP	Ņ	/IGCF		SIP
	IAM	<b>→</b>		<b>→</b>	INVITE
	CASE A				
	ACM	<b>←</b>		←	180 Ringing
				←	200 OK INVITE
				<b>→</b>	ACK
			T <sub>TIR1</sub>		
	ANM	<b>←</b>			
	CASE B				
	ACM	<del>(</del>		<b>←</b>	183 Session Progress
				<b>←</b>	200 OK INVITE
				<b>→</b>	ACK
			T <sub>TIR1</sub>		
	ANM	<del>(</del>			
	CASE C				
				<b>←</b>	200 OK INVITE
				<b>→</b>	ACK
			T <sub>TIR1</sub>		
	CON	<b>←</b>			
			versation		
	REL	<b>→</b>		<b>→</b>	BYE
	RLC	<b>←</b>		<del>-</del>	200 OK BYE

	Values for test purpose TP606006					
VA PRIV_VALUE						
VA_1	ld					
VA_2	User					
VA_3	Header					

## 6.3.2.7 Sub-addressing (SUB)

TP607001	CW Refer		Selection criteria: PICS 5/8		
TSS reference:	ES 283 027 [1], classical is up-sip/ss/sub/	ause 4.7.4.5.2	PICS 5/6		
	130F-31F/33/30B/				
Preconditions:					
Test purpose:	The calling party subado	lress is mapped i	n 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	INVITE:				
values:	P-Asserted-Identity:	sip: user part; isu	b= <subaddress>@hostportion</subaddress>		
ISUP Parameter values:	IAM: ATP(Calling party s	subaddress)			
Comments:	ISUP	MGCF	SIP		
	IAM(ATP)	<b>→</b>	→ INVITE		
	,		← 100 Trying		
	ACM	<b>←</b>	€ 180 Ringing		
	, tem	•	2 100 Kinging		
	ANM	<b>←</b>	← 200 OK INVITE		
			→ ACK		
		Cor	nmunication		
	REL	<b>→</b>	→ BYE		
	RLC	<b>←</b>	← 200 OK BYE		

TP607002	CW Reference ES 283 027 [1], clause		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					
	interworked in the isub parar	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 pa	art; isub= <subado< th=""><th>dress&gt;@hostportion</th></subado<>	dress>@hostportion			
ISUP Parameter values:	IAM: ATP(Called party subac	ddress)				
Comments:	ISUP	MGCF	SIP			
	IAM(ATP)	<b>→</b>	→ INVITE			
			← 100 Trying			
	ACM	<b>←</b>	← 180 Ringing			
	ANM	<b>←</b>	← 200 OK INVITE			
	→ ACK					
		Commu	nication			
	REL	<b>→</b>	→ BYE			
	RLC	+	← 200 OK BYE			

	CW Reference	:	Selection criteria:				
TP607003	ES 283 027 [1], clause	4.7.4.5.2	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 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</subaddress>						
ISUP Parameter	IAM: oFCi: connected line req		•				
values:	ANM: ATP(Connected subado	dress)					
Comments:	ISUP	MGCF	SIP				
	IAM	<b>→</b>	→ INVITE				
			← 100 Trying				
	ACM	<b>←</b>	← 180 Ringing				
			← 200 OK INVITE → ACK				
	ANM( ATP)	<b>←</b>	← UPDATE				
			→ 200 OK UPDATE				
		Commi	unication				
	REL	<b>→</b>	→ BYE				
	RLC	<b>←</b>	← 200 OK BYE				

TP607004	CW Refe ES 283 027 [1], (		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				
	1	parameter of the Fro	the ATP parameter of the received IAM is not om header in the sent INVITE, if the Type of		
SIP Parameter	INVITE:				
values:	P-Asserted-Identity	: sip: user part; isub=	= <subaddress>@hostportion</subaddress>		
ISUP Parameter values:	IAM: ATP(no Calling pa	arty subaddress)			
Comments:	ISUP	MGCF	SIP		
	IAM(ATP)	<b>→</b>	→ INVITE		
			← 100 Trying		
	ACM	<b>←</b>	← 180 Ringing		
	ANM	<b>←</b>	← 200 OK INVITE		
			→ ACK		
		Comr	nunication		
	REL	<b>→</b>	→ BYE		
	RLC	+	← 200 OK BYE		

TP607005		ference: clause 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</subaddress>			
ISUP Parameter values:	IAM: ATP(no Called p	arty subaddress)		
Comments:	ISUP	MGCF	SIP	
	IAM(ATP)	<b>→</b>	→ INVITE	
			← 100 Trying	
	ACM	<b>←</b>	← 180 Ringing	
	ANM	<b>←</b>	← 200 OK INVITE	
			→ ACK	
		Com	munication	
	REL	<b>→</b>	→ BYE	
	RLC	<b>←</b>	← 200 OK BYE	

# 6.3.2.8 Closed user group (CUG)

TP608001	SIP reference	ce: RFC 3261 [6]	ES	ISUP reference: 283 027 [1], clause 7.4.16
TSS reference:	ISUP-SIP/SS/CUG/			
SIP selection	NOT PICS 5/7			
criteria:				
ISUP selection				
criteria:				
Test purpose:	SIP network does 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	No mapping			
values:				
ISUP Parameter values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>	→	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
	Ringing tone			
	ANM	<del>(</del>	<b>←</b>	200 OK INVITE
			→	ACK
		Conve	ersation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	<del>-</del>	<b>←</b>	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/	20 200 021 [1], oldado 111110
SIP selection criteria:	NOT PICS 5/7	
ISUP selection criteria:		
Test purpose:	SIP network does not support CUG, CUG with Ensure that the SUT if an IAM is received with indicator coded as "CUG call without outgo is sent. No INVITE is sent into the SIP netwo	h Optional forward call indicator, CUG call ing access" and CUG interlock code, a REL
SIP Parameter values:	No action	
ISUP Parameter values:	REL: Cause #29	
Comments:	ISUP/BICC SUT  IAM →  REL ←  RLC →	SIP

TSS reference: ISUP-SIP/SS/CUG/ SIP selection criteria:  ISUP selection criteria:  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 value '10' is mapped into <cug> &lt; cugCommunicationIndicator&gt;, the Closed user group interlock cod Parameter Network identity is mapped into <cug> <networkindicator> and the Binary code is mapped into the <cug> <cuginterlockbinarycode>.  SIP Parameter  INVITE:</cuginterlockbinarycode></cug></networkindicator></cug></cug>	e		
criteria:  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 value '10' is mapped into <cug> &lt; cugCommunicationIndicator&gt;, the Closed user group interlock cod Parameter Network identity is mapped into <cug> <networkindicator> and the Binary code is mapped into the <cug> <cuginterlockbinarycode>.  SIP Parameter  INVITE:</cuginterlockbinarycode></cug></networkindicator></cug></cug>	e		
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 value '10' is mapped into <cug> &lt; cugCommunicationIndicator&gt;, the Closed user group interlock cod Parameter Network identity is mapped into <cug> <networkindicator> and the Binary code is mapped into the <cug> <cuginterlockbinarycode>.  SIP Parameter  INVITE:</cuginterlockbinarycode></cug></networkindicator></cug></cug>	e		
Test purpose:  SIP network supports CUG. CUG call indicator value "10" received.  Ensure that Optional Forward Call Indicator Parameter CUG call indicator value '10' is mapped into <cug> &lt; cugCommunicationIndicator&gt;, the Closed user group interlock cod Parameter Network identity is mapped into <cug> <networkindicator> and the Binary code is mapped into the <cug> <cuginterlockbinarycode>.  SIP Parameter  INVITE:</cuginterlockbinarycode></cug></networkindicator></cug></cug>	е		
Test purpose:  SIP network supports CUG. CUG call indicator value "10" received.  Ensure that Optional Forward Call Indicator Parameter CUG call indicator value '10' is mapped into <cug> &lt; cugCommunicationIndicator&gt;, the Closed user group interlock cod Parameter Network identity is mapped into <cug> <networkindicator> and the Binary code is mapped into the <cug> <cuginterlockbinarycode>.  SIP Parameter  INVITE:</cuginterlockbinarycode></cug></networkindicator></cug></cug>	e		
Ensure that Optional Forward Call Indicator Parameter CUG call indicator value '10' is mapped into <cug> &lt; cugCommunicationIndicator&gt;, the Closed user group interlock cod Parameter Network identity is mapped into <cug> <networkindicator> and the Binary code is mapped into the <cug> <cuginterlockbinarycode>.  SIP Parameter  INVITE:</cuginterlockbinarycode></cug></networkindicator></cug></cug>	e		
mapped into <cug> &lt; cugCommunicationIndicator&gt;, the Closed user group interlock cod Parameter <b>Network identity</b> is mapped into <cug> <networkindicator> and the <b>Binary code</b> is mapped into the <cug> <cuginterlockbinarycode>.  SIP Parameter  INVITE:</cuginterlockbinarycode></cug></networkindicator></cug></cug>	e		
Parameter Network identity is mapped into <cug> <networkindicator> and the Binary code is mapped into the <cug> <cuginterlockbinarycode>.  SIP Parameter  INVITE:</cuginterlockbinarycode></cug></networkindicator></cug>			
code is mapped into the <cug> <cuginterlockbinarycode>.  SIP Parameter INVITE:</cuginterlockbinarycode></cug>			
SIP Parameter INVITE:			
values:  cug>	<cug></cug>		
<pre><networkindicator>[derived from IAM Network identity]</networkindicator></pre>			
<pre><cuginterlockbinarycode>[derived from IAM Binary code]</cuginterlockbinarycode></pre> /cugInterlockBinaryCode			
	<pre><cugcommunicationindicator>10</cugcommunicationindicator></pre>		
	IAM:		
	Optional Forward Call Indicator CUG call indicator = "10"		
Closed User Group interlock code  Binary code derived from INVITE XML body <cuginterlockbinarycode></cuginterlockbinarycode>			
Network identity derived from INVITE XML body <a href="https://www.networkIndicator">Network identity derived from INVITE XML body <a href="https://www.networkIndicator">networkIndicator</a></a>			
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:		
		ES 283 027 [1], clause 7.4.1.2		
TSS reference:	ISUP-SIP/SS/CUG/			
SIP selection	PICS 5/7			
criteria:				
ISUP selection				
criteria:				
Test purpose:	SIP network supports CUG. CUG call indicator value "11" received.			
	Francisco that Ontional Famourd Call Indicator Parameter CHC and indicator is (44) in			
	Ensure that Optional Forward Call Indicator Parameter <b>CUG call indicator</b> value '11' is			
	mapped into <cug> &lt; cugCommunicationIndicator&gt;, the Closed user group interlock code</cug>			
	Parameter <b>Network identity</b> is mapped into <cug> <networkindicator> and the <b>Binary code</b> is mapped into the <cug> <cuginterlockbinarycode>.</cuginterlockbinarycode></cug></networkindicator></cug>			
SIP Parameter	INVITE:	Billary Code >:		
values:	<pre>cug&gt;</pre>			
values.	<pre><networkindicator>[derived from IAM Network identity]</networkindicator></pre>			
	<pre><cuginterlockbinarycode>[derived from IAM Binary code]</cuginterlockbinarycode></pre> /cugInterlockBinaryCode>			
	<cugcommunicationindicator>11</cugcommunicationindicator>			
ISUP Parameter	IAM:			
values:	Optional Forward Call Indicator CUG call indicator = "11"			
	Closed User Group interlock code			
	Binary code derived from INVITE XML body < cugInterlockBinaryCode>			
	Network identity derived from INVITE XM			
Comments:	ISUP/BICC SUT	SIP		
	IAM →	→ INVITE		
	ACM <b>←</b>	← 180 Ringing		
	Ringing tone			
	ANM <b>←</b>	◆ 200 OK INVITE		
		→ ACK		
	Conve	rsation		
	REL →	→ BYE		
	RLC <b>←</b>	← 200 OK BYE		

# 6.3.2.9 Call diversion (CDIV)

TP609001	SIP refer	rence: RFC 3261 [6]		ISUP reference:
			ES	283 027 [1], clause 7.4.6
TSS reference:	ISUP-SIP/SS/ CDIV /			
SIP selection	NOT PICS 5/12	AND NOT PICS 5/13 AND NO	OT PICS 5/	14 AND NOT PICS 5/15
criteria:				
ISUP selection				
criteria:				
Test purpose:	CDIV parameter	not mapped		
		0.17		
	Ensure that the SUT if the IAM is received with <b>Redirecting number</b> , <b>original called</b>			
	number and redirection information, continue without disrupting the SIP or ISUP			
CID Donomotor	signalling procedure.			
SIP Parameter values:	No mapping			
ISUP Parameter	IAM: Redirecting number, Original called number, Redirection information			
values:	IAM: Redire	ecting number, Onginal called	i number, R	edirection information
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>	→	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
		Ringing tone		
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
			→	ACK
		Conve	rsation	
	REL	<b>→</b>	<b>→</b>	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	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	IAM with Original call number and redirecting	number Presentation allowed received		
	Engure that the CLIT if the IAM is received with <b>Badinesting number</b> , arising a sulfid			
	Ensure that the SUT if the IAM is received with <b>Redirecting number</b> , <b>original called number</b> Presentation allowed and <b>redirection information</b> Presentation allowed, the			
	redirection counter value is "1", an INVITE is			
	Redirecting number is contained in the hi-targ			
	number is contained in the hi-targeted-to-uri in			
	the latest history entry is mapped from the rec			
SIP Parameter	INVITE: History-Info header			
values:	hi-targeted-to-uri Redirecting number; index=			
	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	IAM:			
values:	Redirection information: "call diversion" Redirection counter = 1			
	Redirection counter = 1 Redirecting indicator = 3			
	Redirecting reason = ISUP_RR			
	Original called number			
	Presentation restriction: Presentation allowed			
	Redirecting number			
	Presentation restriction: Presentation allowed	IT. 010		
Comments:	ISUP SU	_		
	IAM →	→ INVITE ← 180 Ringing		
	1 1 2 1 1 1	<ul><li>← 180 Ringing</li><li>← 200 OK INVITE</li></ul>		
	ANM ←	→ ACK		
	Commu	- /		
	REL →	→ BYE		
	RLC	200 OK BYE		
	<b>←</b>	200 OK BTE		
	•	<u> </u>		

IAM		INVITE	
ISUP Parameter	Source value of parameter	SIP component	Derived value of
	field		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	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	IAM with Original call number Presentation restricted and redirecting number Presentation allowed received			
	Ensure that the SUT if the IAM is received with <b>Redirecting number</b> , <b>original called number</b> Presentation restricted and <b>redirection information</b> Presentation allowed, the redirection counter value is "1", an INVITE is sent containing a History-Info header. The Redirecting number is contained in the hi-targeted-to-uri in index 1, the called party number is contained in the hi-targeted-to-uri in index 1.1. The cause parameter value in the latest history entry is mapped from the redirection reason indicator.			
SIP Parameter	INVITE: History-Info header			
values:	hi-targeted-to-uri Redirecting number?Privacy			
ISUP Parameter	hi-targeted-to uri diverted to user; cause= <b>Cause_value</b> ; index=1.1			
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 SU	IT SIP		
	IAM →	→ INVITE		
	ACM <b>←</b>	<ul><li>180 Ringing</li></ul>		
	ANM <b>←</b>	← 200 OK INVITE		
	_	→ ACK		
	Commu			
	REL →	→ BYE		
	RLC	200 OK BYE		
	<b>←</b>	+		

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	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	IAM with Original call number Presentation allowed and redirecting number Presentation restricted received			
	Ensure that the SUT if the IAM is received with <b>Redirecting number</b> presentation restricted, <b>original called number</b> Presentation allowed and <b>redirection information</b> Presentation restricted, the redirection counter value is "1", an INVITE is sent containing a History-Info header. The Redirecting number is contained in the hi-targeted-to-uri in index 1, the called party number is contained in the hi-targeted-to-uri in index 1.1. The cause parameter value in the latest history entry is mapped from the redirection reason indicator.			
SIP Parameter	INVITE: History-Info header			
values:	hi-targeted-to-uri Redirecting number?Privacy=history; index=1,			
10110 0	hi-targeted-to uri diverted to user; cause=Cause_value; index=1.1			
ISUP Parameter values:	IAM: Redirection information: "call diversion"			
values:		Redirection counter = 1		
	Redirecting indicator = 4			
	Redirecting reason = ISUP_RR			
	Original called number			
	Presentation restriction: Presentation allowed			
	Padiracting number			
	Redirecting number Presentation restriction: Presentation restricted			
Comments:	ISUP SU			
	IAM →	→ INVITE		
	ACM ←	← 180 Ringing		
	ANM ←	← 200 OK INVITE		
		→ ACK		
	Commu			
	REL →	→ BYE		
	RLC 200 OK BYE			
	<b>←</b>	+		

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	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	IAM with Original call number Presentation all restricted received	owed and redirecting number Presentation		
	Ensure that the SUT if the IAM is received with <b>Redirecting number</b> presentation restricted, <b>original called number</b> Presentation restricted and <b>redirection information</b> Presentation restricted, the redirection counter value is "1", an INVITE is sent containing a History-Info header. The Redirecting number is contained in the hi-targeted-to-uri in index 1, the called party number is contained in the hi-targeted-to-uri in index 1.1. The cause parameter value in the latest history entry is mapped from the redirection reason indicator.			
SIP Parameter	INVITE: History-Info header			
values:	hi-targeted-to-uri Redirecting number?Privacy			
10115 5	hi-targeted-to uri diverted to user; cause=Cau	se_value; index=1.1		
ISUP Parameter	IAM:			
values:	Redirection information: "call diversion" Redirection counter = 1			
	Redirecting indicator = 4			
	Redirecting reason = ISUP_RR			
	Original called number			
	Presentation restriction: Presentation restricted			
	Radiracting number			
	Redirecting number Presentation restriction: Presentation restricte	4		
Comments:	ISUP SU			
	IAM →	→ INVITE		
	ACM ←	← 180 Ringing		
	ANM ←	← 200 OK INVITE		
		→ ACK		
	Commu	nication		
	REL →	→ BYE		
	RLC	200 OK BYE		
	+	<b>←</b>		

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]	-	SUP reference: 3 027 [1], clause 7.5.4
TSS reference:	ISUP-SIP/SS/ CDIV /		5 62. [1], e.u.u.ce 1161.
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15		
criteria:			
ISUP selection			
criteria:			
Test purpose:	IAM with Original call number and redirecting Redirecting indicator indicates "all redirection"		
	Ensure that the SUT if the IAM is received with 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 hi-targeted-to-uri in index 1, the called party number is contained in the hi-targeted-to-uri in index 1.1. The cause parameter value in the latest history entry is mapped from the redirection reason indicator.		
SIP Parameter	INVITE: History-Info header		
values:	hi-targeted-to-uri Redirecting number?Privacy hi-targeted-to uri diverted to user; cause= <b>Cau</b> :		
ISUP Parameter	IAM:		
values:	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 SU	T	SIP
	IAM →	<b>→</b>	INVITE
	ACM ←	<b>←</b>	180 Ringing
	ANM ←	<del>(</del>	200 OK INVITE
	_	<b>→</b>	ACK
	Commu		D)/E
	REL →	<b>→</b>	BYE
	RLC		200 OK BYE
	<b>+</b>	+	

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]		IP reference:	
		ES 283 0	27 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/ CDIV /			
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15			
criteria:				
ISUP selection				
criteria:	IAM with Original call number and radirecting	number Dresented	tion allowed received	
Test purpose:	IAM with Original call number and redirecting Redirection counter value 2	iumber Presentat	ion allowed received,	
	Redirection counter value 2			
	Ensure that the SUT if the IAM is received with	Redirecting nu	mber Presentation	
	allowed, original called number Presentation			
	Presentation allowed, the redirection counter v			
	History-Info header. The Original called number			
	index 1. The Redirecting number is contained			
	called party number is contained in the hi-targ			
SIP Parameter	parameter value in the latest history entry is m	apped from the re	edirection reason indicator.	
values:	INVITE: History-Info header hi-targeted-to-uri Original called number; index	-1		
values.	hi-targeted-to-uri Redirecting number; cause=			
	hi-targeted-to-uri called party number; cause=		dex=1.1.1	
ISUP Parameter	IAM:			
values:	Redirection information: "call diversion"			
	Redirection counter = 2			
	Redirecting indicator = 3			
	Redirecting reason = ISUP_RR			
	Original called number Presentation restriction: Presentation allowed			
	Presentation restriction. Presentation allowed			
	Redirecting number			
	Presentation restriction: Presentation allowed			
Comments:	ISUP SU	T S	SIP	
	IAM →	<b>→</b> IN	NVITE	
	ACM ←	<b>←</b> 1	80 Ringing	
	ANM ←	<b>←</b> 2	00 OK INVITE	
		<b>→</b> A	CK	
	Commu			
	REL →		SYE	
	RLC	2	00 OK BYE	
	<b>+</b>	+		

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

TP609008	SIP reference: RFC 3261 [6]		ISUP reference:		
		ES 28	3 027 [1], clause 7.5.4		
TSS reference:	ISUP-SIP/SS/ CDIV /				
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15				
criteria:					
ISUP selection					
criteria:					
Test purpose:	IAM with Original call number Presentation resallowed received. Redirection counter value 2		directing number Presentation		
	anowed received, redirection counter value 2				
	Ensure that the SUT if the IAM is received with				
	allowed, original called number Presentation				
	Presentation allowed, the redirection counter v				
	History-Info header. The Original called number				
	index 1. The Redirecting number is contained				
	called party number is contained in the hi-targ				
SIP Parameter	parameter value in the latest history entry is m INVITE: History-Info header	iapped from th	e redirection reason indicator.		
values:	hi-targeted-to-uri Original called number?Priva	ov-history inc	tev-1		
values.	hi-targeted-to-uri Redirecting number; cause=				
	hi-targeted-to-uri called party number; cause=				
ISUP Parameter	IAM:		,		
values:	Redirection information: "call diversion"				
	Redirection counter = 2				
	Redirecting indicator = 3				
	Redirecting reason = ISUP_RR	Redirecting reason = ISUP_RR			
	Original called number				
	Presentation restriction: Presentation restricted	d			
	Redirecting number				
	Redirecting number Presentation restriction: Presentation allowed				
Comments:	ISUP SU	IT	SIP		
Joinnetts.	IAM →	,, →	INVITE		
	ACM +	<b>*</b>	180 Ringing		
	ANM ←	÷	200 OK INVITE		
		÷	ACK		
	Commu	=	,		
	REL →	→	BYE		
	RLC	-	200 OK BYE		
	<b>←</b>	<b>←</b>			

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		487
	'0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP609009	SIP reference: RFC 3261 [6]	-	SUP reference:	
		ES 283	3 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/ CDIV /			
SIP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AND PICS 5/15			
criteria:				
ISUP selection				
criteria:				
Test purpose:	IAM with Original call number Presentation all		ecting number Presentation	
	restricted received, Redirection counter value	2		
	Ensure that the SUT if the IAM is received with	Pedirecting	number Presentation	
	restricted, original called number Presentation			
	Presentation restricted, the redirection counter			
	History-Info header. The Original called number			
	index 1. The Redirecting number is contained			
	called party number is contained in the hi-targe	eted-to-uri in in	dex 1.1.1. The cause	
	parameter value in the latest history entry is m	apped from the	e redirection reason indicator.	
SIP Parameter	INVITE: History-Info header			
values:	hi-targeted-to-uri Original called number; index			
	hi-targeted-to-uri Redirecting number; ?Privac			
IOUD D	hi-targeted-to-uri called party number; cause=	Cause_value;	Index=1.1.1	
ISUP Parameter values:	IAM: Redirection information: "call diversion"			
values.	Redirection counter = 2			
	Redirecting indicator = 4			
	Redirecting reason = ISUP_RR			
	Troullouing Todoon = 1001 _ttt			
	Original called number			
	Presentation restriction: Presentation allowed			
	Redirecting number			
Comments:	Presentation restriction: Presentation restricted		ein	
Comments:	ISUP SU	· →	SIP	
	IAM → ACM ←	-	INVITE	
		<b>+</b>	180 Ringing	
	ANM ←	<b>→</b>	200 OK INVITE ACK	
	Commu	=	AUR	
	REL →	iication →	BYE	
	RLC	7	200 OK BYE	
	<b>←</b>	<b>←</b>	ZOO ON BIE	
	_			

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		487
	'0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP609010	SIP reference: RFC 3261 [6]		SUP reference: 027 [1], clause 7.5.4
TSS reference:	ISUP-SIP/SS/ CDIV /	LO 203	027 [1], clause 7.5.4
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 restricted, <b>original called number</b> Presentation Presentation restricted, the redirection counter History-Info header. The Original called numbindex 1. The Redirecting number is contained called party number is contained in the hi-targ parameter value in the latest history entry is more than the second	on restricted and restricted and restricted and restricted in the history and hist	d redirection information INVITE is sent containing a in the hi-tatgeted-to uri in the ed-to-uri in index 1.1, the dex 1.1.1. The cause
SIP Parameter	INVITE: History-Info header		
values:	hi-targeted-to-uri Original called number?Priva		
	hi-targeted-to-uri Redirecting number; ?Privac		
ISUP Parameter	hi-targeted-to-uri called party number; cause=	Cause_value;	Index=1.1.1
values:	Redirection information: "call diversion"		
values.	Redirection counter = 2		
	Redirecting indicator = 4		
	Redirecting reason = ISUP_RR		
	<del></del>		
	Original called number		
	Presentation restriction: Presentation restricte	d	
	De dine etia a accusab e a		
	Redirecting number Presentation restriction: Presentation restricte	4	
Comments:	ISUP SU		SIP
Commonto.	IAM →	· →	INVITE
	ACM	<del>-</del>	180 Ringing
	ANM	<del>-</del>	200 OK INVITE
		<b>→</b>	ACK
	Commu	nication	
	REL →	<b>→</b>	BYE
	RLC		200 OK BYE
	<b>←</b>	<b>←</b>	

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		487
	'0100'B		
	Deflection immediate		480
	response '0101'B		
	Mobile subscriber not		503
	reachable		

TP609011	SIP reference: RFC 3261 [6]	ISUP reference:	F 4
T00 (	IOUE OUD/OO/ ODIN//	ES 283 027 [1], clause 7	.5.4
TSS reference: SIP selection	ISUP-SIP/SS/ CDIV /	ID DIOC 5/45	
SIP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ID PICS 5/15	
ISUP selection			
criteria:			
Test purpose:	IAM with Original call number and redirecting	number received. Redirection cou	nter value
	3		
	Ensure that the SUT if the IAM is received with	Redirecting number, original o	alled
	number and redirection information, the red		
	sent containing a History-Info header. The Ori		
	tatgeted-to uri in the index 1. The Redirecting		
	in index 1.1.1, the called party number is contained.		
	1.1.1.1. The cause parameter value in the late	st history entry is mapped from th	е
SIP Parameter	redirection reason indicator. INVITE: History-Info header		
values:	hi-targeted-to-uri Original called number; inde	·=1	
values.	hi-targeted-to-uri Dummy entry(PIXIT); cause= <b>302</b> ; index=1.1,		
	hi-targeted-to-uri Redirecting number; cause= <b>486</b> ; index=1.1.1,		
	hi-targeted-to-uri called party number; cause=	Cause_value; index=1.1.1.1	
ISUP Parameter	IAM:		
values:	Redirection information: "call diversion"		
	Redirection counter = 3		
	Redirecting reason = ISUP_RR		
	Original called number		
	Redirecting number		
Comments:	ISUP SU		
	IAM →	→ INVITE	
	ACM ←	← 180 Ringing	
	ANM ←	← 200 OK INVITE → ACK	
	Commu	- /1011	
	REL →	→ BYE	
	RLC	200 OK BYE	
	<b>←</b>	200 OK B1E ←	
		•	

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

TP609012	SIP reference: RFC 3261 [6]	ISUP reference:	
TCC voferonce:	ICUID CID/CC/ CDIV/	ES 283 027 [1], clause 7.5.4	
TSS reference: SIP selection	ISUP-SIP/SS/ CDIV / PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ID DICC F/AF	
criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ND 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 <b>Redirecting number</b> , <b>original called number</b> Presentation allowed and <b>redirection information</b> 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 hi-targeted-to-uri in index 1.1, the called party number is contained in the hi-targeted-to-uri in index 1.1.1. The cause parameter value in the latest history entry is mapped from the redirection reason indicator.		
SIP Parameter	INVITE: History-Info header		
values:	hi-targeted-to-uri Original called number?Privacy=history; index=1,		
	hi-targeted-to-uri Redirecting number?Privacy=history; cause= <b>302</b> ; index=1.1,		
	hi-targeted-to-uri called party number; cause=	Cause_value; index=1.1.1	
ISUP Parameter	IAM:		
values:	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 SU	T SIP	
	IAM →	→ INVITE	
	ACM +	← 180 Ringing	
	ANM	€ 200 OK INVITE	
		→ ACK	
	Commui		
	REL →	<b>→</b> BYE	
	RLC	200 OK BYE	
	<b>←</b>	+	

IAM		INVITE	
ISUP Parameter or	Source value of parameter	SIP component	Derived value of
	field		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

TP609013	SIP reference: RFC 3261 [6]	ISUP reference:		
TCC reference:	ICLID CID/CC/CDIV/	ES 283 027 [1], clause 7.5.4		
TSS reference: SIP selection	ISUP-SIP/SS/CDIV/			
criteria:				
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	UD DICC E/AE		
criteria:	PICS 5/12 AIND PICS 5/13 AIND PICS 5/14 AI	ND PICS 5/15		
Test purpose:	181 Received, Notification subscription opt	tion according the Privacy header in the		
rest purpose.	History-Info header			
	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 in 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	Notification subscription option = ISUP_NSO		
	and the <b>Generic notification</b> indicator parameter = call is diverting			
SIP Parameter	181: History-Info header			
values:	hi-targeted-to-uri Redirecting number; index=1	,		
	hi-targeted-to uri diverted to user; cause=Caus	se value?Privacy= <b>priv-value</b> ; index=1.1		
ISUP Parameter	ACM			
values:	BCI: No indication (00),			
	GenNot: Call is diverting (1111011),			
	Call diversion Info: ISUP_NSO			
	Redirection number: derived from the Hi-targe			
Comments:	ISUP SU	IT SIP		
	IAM →	→ INVITE		
	ACM <b>←</b>	<ul> <li>181 Being Forwarded</li> </ul>		
	CPG ←	← 180 Ringing		
	ANM ←	← 200 OK INVITE		
		→ ACK		
	Commu	nication		
	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

TP609014	SIP reference: RFC 3261 [6]	ISUP reference:		
TSS reference:	ISUP-SIP/SS/CDIV/	ES 283 027 [1], clause 7.5.4		
SIP selection	1301 -311 /33/0010/			
criteria:				
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AI	ND PICS 5/15		
criteria:				
Test purpose:	181 received, ACM no indication is sent: NSC	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 se	t to the hist-info element concerning the		
	redirecting uri and the diverted-to-uri then			
	Sends of an ACM massage indicating a first d	Condo of an ACM manage indicating a first diversion with the Backward call in directors		
	Sends of an <b>ACM</b> message indicating a first diversion with the <b>Backward call indicators</b> 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			
010.0	and the Generic notification indicator parameter = call is diverting			
SIP Parameter	181: History-Info header hi-targeted-to-uri Redirecting number? <b>Privacy=history</b> ; index=1,			
values:	hi-targeted-to uri diverted to user; cause=Cau			
ISUP Parameter	ACM	se value: Filvacy=Ilistory, Ilidex=1.1		
values:	BCI: No indication (00),			
1414551	GenNot: Call is diverting (1111011),			
	Call diversion Info: presentation not allowed			
	Redirection number: derived from the Hi-targe	t-to-uri of the last History-Info entry		
Comments:	ISUP SU	JT SIP		
	IAM →	→ INVITE		
	ACM <b>←</b>	<ul> <li>181 Being Forwarded</li> </ul>		
	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	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	D PICS 5/15	
criteria:			
Test purpose:	181 received sending of <b>Redirection number</b>	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	181: History-Info header	•	
values:	hi-targeted-to-uri Redirecting number; index=1		
	hi-targeted-to uri diverted to user; cause=Caus	e value?Privacy= <b>priv-value</b> ; index=1	.1
ISUP Parameter	ACM:		
values:	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 SU		
	IAM →	→ INVITE	
	ACM +	<ul> <li>181 Being Forwarded</li> </ul>	
	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	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ID PICS 5/15
criteria:		
Test purpose:	181 received, coding of notification subscrip	tion option in the CPG Progress
	Ensure that the SUT, when an ACM has been Forwarded) containing the History-Index, Priva Info header, Privacy, priv-value component co Sends a CPG message indicating a first di parameter coded Event indicator = PROGRESS, Redirection number:  Redirection number: derived from the Hi-targe the Call diversion information parameter Notification subscription option = ISUP_NSO and the Generic notification indicator parameter	acy, priv-value component and the History- ncerning the diverted-to uri version with the <b>Event information</b> t-to-uri of the last History-Info entry
SIP Parameter	181: History-Info header	Ü
values:	hi-targeted-to-uri Redirecting number; index=1	,
	hi-targeted-to uri diverted to user; cause=Caus	se value?Privacy= <b>Priv-value</b> ; index=1.1
ISUP Parameter	CPG:	
values:	Event indicator = PROGRESS, GenNot: Call is diverting (1111011), Call diversion Info: ISUP_NSO Redirection number: derived from the Hi-targe	t-to-uri of the last History-Info entry
Comments:	ISUP SU	T SIP
	IAM →	→ INVITE
	ACM ←	← 180 Ringing
	CPG +	<ul> <li>181 Being Forwarded</li> </ul>
	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		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,

SIP selection criteria:	UP-SIP/SS/CDIV/ CS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ES 283 027 [1], clause 7.5.4	
SIP selection criteria:			
	CS 5/12 AND PICS 5/13 AND PICS 5/14 AN		
ISUP selection PI	CS 5/12 AND PICS 5/13 AND PICS 5/14 AN		
		ND PICS 5/15	
criteria:			
	181 received Privacy=history concerning the redirecting and the diverted-to URI setting of NSO in the CPG Progress		
Fo	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 <b>Event information parameter</b> coded		
	Event indicator = PROGRESS,		
	Redirection number:		
	Redirection number: derived from the Hi-target-to-uri of the last History-Info entry the <b>Call diversion information parameter</b>		
	Notification subscription option = presentation not allowed		
an	the <b>Generic notification indicator paran</b>	neter coded	
	otification indicator = call is diverting,		
	1: History-Info header		
	targeted-to-uri Redirecting number?Privacy		
	targeted-to uri diverted to user; cause=Caus	se value? <b>Privacy=history</b> ; index=1.1	
	PG:		
	vent indicator = PROGRESS,		
	enNot: Call is diverting (1111011), all diversion Info: presentation not allowed		
	UP SU	T SIP	
IA		→ INVITE	
	CM <del>C</del>	€ 180 Ringing	
	PG ←	← 181 Being Forwarded	
	√ M <del>←</del>	€ 200 OK INVITE	
/ "	•	→ ACK	
	Commur		
RE		→ BYE	
RL		€ 200 OK BYE	

TP609018	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4
TSS reference:	ISUP-SIP/SS/CDIV/	203 027 [1], clause 7.3.4
SIP selection	1001 -011 /00/0011/	
criteria:		
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ND PICS 5/15
criteria:		
Test purpose:	181 received setting of Redirection number i	restriction in the CPG Progress
	Ensure that the SUT, when an ACM has been	
	Forwarded) containing the History-Info heade	
	Sends a CPG message indicating a first di	version with the <b>Event information</b>
	parameter coded	
	Event indicator = PROGRESS, Redirection number:	
	Redirection number: Redirection number: derived from the Hi-targe	t to uri of the last History Info entry
	Redirection number restriction indicator= I	
	and the Generic notification indicator param	<del>_</del>
SIP Parameter	181: History-Info header	g
values:	hi-targeted-to-uri Redirecting number; index=1	,
	hi-targeted-to uri diverted to user; cause=Caus	
ISUP Parameter	CPG:	
values:	Event indicator = PROGRESS,	
	GenNot: Call is diverting (1111011),	
	Redirection number: ISUP_ReNr	
	Redirection number restriction indicator: ISUP	
Comments:	ISUP SU	_
	IAM -	→ INVITE
	ACM ←	← 180 Ringing
	CPG ←	← 181 Being Forwarded
	ANM ←	€ 200 OK INVITE
	_	→ ACK
	Commu	
	REL →	→ BYE
	RLC <b>←</b>	← 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]		SUP reference: 3 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/CDIV/	20 20	5 027 [1]; 01dd00 71014	
SIP selection				
criteria:				
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ND PICS 5/15		
criteria:				
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 the Call diversion information parameter  Notification subscription option = ISUP_NSO  and the Generic notification indicator parameter = call is diverting			
SIP Parameter	180: History-Info header			
values:	hi-targeted-to-uri Redirecting number; index=1, hi-targeted-to uri diverted to user; cause=Cause value?Privacy= <b>priv-value</b> ; index=1.1			
ISUP Parameter	ni-targeted-to uri diverted to user; cause=Cause	se value?Priva	cy= <b>priv-value</b> ; index=1.1	
values:	Event indicator = ALERTING,			
valuoo.	GenNot: Call is diverting (1111011),			
	Redirection number: derived from the Hi-targe	t-to-uri of the la	ast History-Info entry	
	Notification subscription option: ISUP_NSO			
Comments:	ISUP SU		SIP	
	IAM →	<b>→</b>	INVITE	
	ACM ← CPG ←	<b>+</b>	181 Being Forwarded	
	ANM +	<del>-</del>	180 Ringing 200 OK INVITE	
	AINIVI	<b>→</b>	ACK	
	Commu	=	AON	
	REL →		BYE	
	RLC ←	<del>-</del>	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	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ND PICS 5/15		
criteria:	100 1 1000 11 11 11 11 11	1100		
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:  Number digits derived from the Hi-target-to-uri of the last History-Info entry  The Call diversion information parameter  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? <b>Privacy</b>			
	hi-targeted-to uri diverted to user; cause=Caus	se value? <b>Privacy=history</b> ; index=1.1		
ISUP Parameter	CPG:			
values:	Event indicator = ALERTING,			
	GenNot: Call is diverting (1111011), Redirection number: derived from the Hi-targe	t-to-uri of the last History-Info entry		
	Notification subscription option: Presentation r			
Comments:	ISUP SU			
	IAM →	→ INVITE		
	ACM <b>←</b>	<ul> <li>181 Being Forwarded</li> </ul>		
	CPG ←	← 180 Ringing		
	ANM ←	€ 200 OK INVITE		
		→ ACK		
	Commu			
	REL →	→ BYE		
	RLC ←	← 200 OK BYE		

SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.5.4	
ISUP-SIP/SS/CDIV/		
PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ID PICS 5/15	
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:  Number digits derived from the Hi-target-to-uri of the last History-Info entry  Redirection number restriction parameter  Redirection number restriction indicator = ISUP_ReNrReIn  and the Generic notification indicator coded		
180: History-Info header		
,	se value?Privacy= <b>priv-value</b> ; index=1.1	
Event indicator = ALERTING,	PoNrPolo	
	→ INVITE	
	← 181 Being Forwarded	
	← 180 Ringing	
	€ 200 OK INVITE	
7 11 11 11	→ ACK	
Commu	2 7.611	
	→ BYE	
	€ 200 OK BYE	
	ISUP-SIP/SS/CDIV/  PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN  180 received, CPG Alerting is sent, setting of a sent that the SUT on receipt of 180 (Ring received before) containing the History-Information concerning diverted-to urith Sends a CPG message indicating a first disparameter coded Event indicator = ALERTING, Redirection number:  Number digits derived from the Hi-target-to-urith Redirection number restriction parameter Redirection number restriction indicator = ISUI and the Generic notification indicator coded Notification indicator = call is diverting  180: History-Info header hi-targeted-to-urith Redirecting number; index=1 hi-targeted-to-urith diverted to user; cause=Cause CPG:  Event indicator = ALERTING, Redirection number restriction indicator: ISUP ISUP  ISUP  ISUP  ISUP  SU  Communication  Communication	

	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: 3 027 [1], clause 7.5.4
TSS reference:	ISUP-SIP/SS/CDIV/		
SIP selection			
criteria:			
ISUP selection	PICS 5/12 AND PICS 5/13 AND PICS 5/14 AN	ND PICS 5/15	
criteria:			
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	200: History-Info header		
values:	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	ANM:		
values:	Redirection number restriction indicator: ISUP	_	
Comments:	ISUP SU	IT	SIP
	IAM →	<b>→</b>	INVITE
	ACM ←	<b>←</b>	181 Being Forwarded
	CPG ←	+	180 Ringing
	ANM ←	<b>←</b>	200 OK INVITE
		<b>→</b>	ACK
	REL →	<b>→</b>	BYE
	RLC <b>←</b>	+	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 28	3 027 [1], clause 7.5.4
TSS reference:	ISUP-SIP/SS/CDIV/			
SIP selection	PICS 10/6			
criteria:				
ISUP selection criteria:	PICS 5/12 AND PICS	5/13 AND PICS 5/14 AI	ND PICS 5/15	
Test purpose:	181 Received, no map	pping 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	181: History-Info header			
values:		recting number; index=1		
	hi-targeted-to uri diver	ted to user; cause=Cau	se value; index	x=1.1
ISUP Parameter				
values:				
Comments:	ISUP	SU	· <del>-</del>	SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
			<b>←</b>	181 Being Forwarded
	ACM	<b>←</b>	<b>←</b>	180 Ringing
	ANM	<del>(</del>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Commu	nication	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	+	+	200 OK BYE

TP609024	SIP reference: RFC 3261 [6]	1	SUP reference: 3 027 [1], clause 7.5.4	
TSS reference:	ISUP-SIP/SS/CDIV/		5 02.7 [1], Slades 7.6.4	
SIP selection criteria:	PICS 10/6			
ISUP selection criteria:	PICS 5/12 AND PICS 5/13 AND PICS 5/14 A	ND 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= hi-targeted-to uri diverted to user; cause=Cau		:=1.1	
ISUP Parameter values:		,		
Comments:	ISUP SI IAM → ACM ←	)T	SIP INVITE 180 Ringing 181 Being Forwarded 200 OK INVITE ACK	
	REL → RLC ←	<b>→</b>	BYE 200 OK BYE	

TP609025	SIP reference: RFC 326	1 [6]		ISUP reference:
TSS reference:	ISUP-SIP/SS/CDIV/		E3 20	3 027 [1], clause 7.5.4
SIP selection	PICS 10/6			
criteria:	1 100 10/0			
ISUP selection criteria:	NOT PICS 5/12 AND NOT PICS	5/13 AND NO	OT PICS 5/14	AND NOT PICS 5/15
Test purpose:	180 received containing History-	Info header, r	no mapping.	
	Ensure that the SUT on receipt received before) containing the mapped			
SIP Parameter	180: History-Info header			
values:	hi-targeted-to-uri Redirecting nur			. 4 4
IOUD Deservator	hi-targeted-to uri diverted to user	; cause=Cau	se value; index	(=1.1
ISUP Parameter values:				
Comments:	ISUP	SU	JT	SIP
	IAM -	<b>→</b>	→	INVITE
			<b>←</b>	181 Being Forwarded
	ACM	<b>-</b>	<b>←</b>	180 Ringing
	ANM	<b>-</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Commu	nication	
	REL -	<b>→</b>	<b>→</b>	BYE
	RLC	<b>-</b>	+	200 OK BYE

TP609026	SIP reference	e: RFC 3261 [6]		ISUP reference: 3 027 [1], clause 7.5.4				
TSS reference:	ISUP-SIP/SS/CDIV/	ISUP-SIP/SS/CDIV/						
SIP selection criteria:	PICS 10/6							
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; index=1.1							
ISUP Parameter values:	No mapping							
Comments:	ISUP IAM ACM ANM	<b>→ ← ←</b>	SUT	SIP INVITE 181 Being Forwarded 180 Ringing 200 OK INVITE ACK				
	REL RLC	<b>→</b>	<b>→</b>	BYE 200 OK BYE				

## 6.3.2.10 User to user signalling (UUS)

TP610001	SIP referen	ce: RFC 3261 [6]	ES	ISUP reference: 283 027 [1], clause 7.4.21
			ITU-T R	ec. Q.737.1 [33], clause 1.1.7
TSS reference:	ISUP-SIP/SS/ UUS	/		
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	by the network  Ensure that the SU service 1 request	T if the IAM is received wit	h <b>User-to-</b> u cator in the	er-to-user information discarded user information as an implicit ACM "UUI discarded by the signalling procedure.
SIP Parameter	No mapping			3 - 31
values:				
ISUP Parameter values:	ACM: User-to-indication".	cator "UUI discarded by the	e network",	Service 1 response "No
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
		Ringing tone		
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conve	rsation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	+	<b>←</b>	200 OK BYE

TP610002	SIP reference: RF	C 3261 [6]		ISUP reference: 283 027 [1], clause 7.4.21 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:	response  Ensure that the SUT if the	IAM is received wit to-user indicator in	h an <b>explic</b> the ACM "S	Service 1 not provided" and		
SIP Parameter	No mapping		gg p			
values:	3 311 3					
ISUP Parameter values:	ACM: User-to-indicator Se provided"	ervice 1 response "l	Not			
Comments:	ISUP/BICC	SUT		SIP		
	IAM	<b>→</b>	→	INVITE		
	ACM	<b>←</b>	<b>←</b>	180 Ringing		
	Ringing tone					
	ANM	<b>←</b>	+	200 OK INVITE		
			<b>→</b>	ACK		
		Conve	rsation			
	REL	<b>→</b>	<b>→</b>	BYE		
	RLC	+	+	200 OK BYE		

TP610003	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [1], clause 7.4.21				
		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 <b>explicit service 1 request</b> "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  IAM →  REL #29 ←  RLC →	SIP				

TP610004	SIP refere	ence: RFC 3261 [6]		ISUP reference: 283 027 [1], clause 7.4.21
			ITU-T	Rec. Q.737 [33], clause 1.2.7
TSS reference:	ISUP-SIP/SS/ UU	IS /		
SIP selection criteria:				
ISUP selection criteria:	PICS 11/1 AND F	PICS 11/2		
Test purpose:	response  Ensure that the S essential" returns	ice 2 explicit request not es UT if the IAM is received w a User-to-user indicator in disrupting the SIP or ISUP:	ith an <b>explic</b> the ACM "Se	ervice 2 not provided" and
SIP Parameter	No mapping		<u> </u>	
values:				
ISUP Parameter values:	ACM: User-to-ind	dicator Service 2 response	"Not provide	d"
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
		Ringing tone		
	ANM	<b>*</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conv	ersation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	<del>(</del>	+	200 OK BYE

TP610005	SIP reference:	RFC 3261 [6]		ISUP reference: ES 283 027 [1], clause 7.4.21 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 <b>explicit service 2 request</b> "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, diag	nostics value	0x2a				
Comments:	ISUP/BICC		SUT	SIP			
	IAM	<b>→</b>					
	REL #29	<b>←</b>					
	RLC	<b>→</b>					

TP610006	SIP reference:	RFC 3261 [6]		ISUP reference:		
				283 027 [1], clause 7.4.21		
			ITU-T Re	ec. 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 1	1/2				
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 <b>explicit service 3 request</b> "Not essential" returns a User-to-user indicator in the ANM "Service 3 not provided" and continue without disrupting the SIP or ISUP signalling procedure.					
SIP Parameter	No mapping					
values:						
ISUP Parameter values:	ACM: User-to-indicator	, Service 3 response	"Not provide	ed"		
Comments:	ISUP/BICC	SUT		SIP		
	IAM	<b>→</b>	→	INVITE		
	ACM	<b>←</b>	<b>←</b>	180 Ringing		
	Ringing tone					
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE		
			<b>→</b>	ACK		
		Conve	ersation			
	REL	<b>→</b>	<b>→</b>	BYE		
	RLC	+	+	200 OK BYE		

TP610007	SIP reference	: RFC 3261 [6	5]	ISUP reference: ES 283 027 [1], clause 7.4.21 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 <b>explicit service 3 request</b> "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	<b>→</b>					
	REL #29	<b>←</b>					
	RLC	<b>→</b>					

TP610008	SIP reference	ce: RFC 3261 [6]		ISUP reference: 283 027 [1], clause 7.4.21 ec. Q.737.1 [33], clause 1.3.7.1
TSS reference:	ISUP-SIP/SS/ UUS	1		
SIP selection criteria:				
ISUP selection criteria:	PICS 11/1 AND PIC	S 11/2		
Test purpose:	rejected by sending	a FRJ if the FAR is received		upported in the confirmed state,
SIP Parameter values:	No action			
ISUP Parameter	FRJ: User-to-user	indicator = "Service 3 r	not provided"	
values:			•	
Comments:	ISUP/BICC	SUT	Γ	SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<del>-</del>	<b>←</b>	180 Ringing
		Ringing tone		
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
			versation	
	FAR	<b>→</b>		
	FRJ	<b>←</b>		
			versation	
	REL	<b>→</b>	→	BYE
	RLC	+	+	200 OK BYE

TP610009	SIP refere	nce: RFC 3261 [6]			ISUP reference: 283 027 [1], clause 7.4.21 IU-T Rec. Q.737.1 [33], clause 1.1.5.2.5.2.2
TSS reference:	ISUP-SIP/SS/ UU	S/			
SIP selection criteria:					
ISUP selection criteria:	NOT PICS 11/2				
Test purpose:	Ensure that the S	UT if the IAM is rece	eived with an <b>ex</b>	plici	t service 1 request "Not nalling procedure. No response
SIP Parameter values:	No mapping				
ISUP Parameter values:					
Comments:	ISUP/BICC		SUT		SIP
	IAM	<b>→</b>		<b>→</b>	INVITE
	ACM	← Ringing tone		<b>←</b>	180 Ringing
	ANM	<b>←</b>		←	200 OK INVITE
				<b>→</b>	ACK
			Conversation		
	REL	<b>→</b>		<b>→</b>	BYE
	RLC	+		<b>←</b>	200 OK BYE

TP610010	SIP refere	nce: RFC 3261 [6]			ISUP reference: 283 027 [1], clause 7.4.21 IU-T Rec. Q.737.1 [33], clause 1.1.5.2.5.2.2
TSS reference:	ISUP-SIP/SS/ UUS	S /			clause 1.1.5.2.5.2.2
SIP selection	1001 -011 /00/ 000	<i>51</i>			
criteria:					
ISUP selection criteria:	NOT PICS 11/2				
Test purpose:	Ensure that the SU		ved with an <b>ex</b>	plici	orted, no response it service 1 request "essential" cedure. No response to this
SIP Parameter	No action				
values:					
ISUP Parameter values:					
Comments:	ISUP/BICC		SUT		SIP
	IAM	<b>→</b>		<b>→</b>	INVITE
	ACM	<b>←</b>		←	180 Ringing
		Ringing tone			
	ANM	<b>←</b>		←	200 OK INVITE
				<b>→</b>	ACK
			Conversation		
	REL	<b>→</b>		<b>→</b>	BYE
	RLC	+		<b>←</b>	200 OK BYE

TP610011	SIP refere	nce: RFC 3261 [6]			ISUP reference: 283 027 [1], clause 7.4.21 IU-T Rec. Q.737.1 [33], clause 1.2.5.2.5.2.1
TSS reference:	ISUP-SIP/SS/ UU	S/			
SIP selection criteria:					
ISUP selection criteria:	NOT PICS 11/2				
Test purpose:	Ensure that the SI	JT if the IAM is rece	eived with an <b>ex</b>	plici	it service 2 request "Not nalling procedure. No response
SIP Parameter	No mapping				
values:					
ISUP Parameter values:					
Comments:	ISUP/BICC		SUT		SIP
	IAM	<b>→</b>		<b>→</b>	INVITE
	ACM	<b>←</b>		←	180 Ringing
		Ringing tone			
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE
				<b>→</b>	ACK
			Conversation		
	REL	<b>→</b>		<b>→</b>	BYE
	RLC	+		<b>←</b>	200 OK BYE

TP610012	SIP refere	nce: RFC 3261 [6]		ISUP reference: 6 283 027 [1], clause 7.4.21 ITU-T Rec. Q.737.1 [33], clause 1.2.5.2.5.2.1				
TSS reference:	ISUP-SIP/SS/ UUS	SUP-SIP/SS/ UUS /						
SIP selection								
criteria:								
ISUP selection	NOT PICS 11/2							
criteria:								
Test purpose:	User-to-user service	ce 2 explicit request e	ssential not sup	ported, no response				
	continue without di request.			cit service 2 request "essential" rocedure. No response to this				
SIP Parameter	No action							
values:								
ISUP Parameter								
values:								
Comments:	ISUP/BICC	S	UT	SIP				
	IAM	<b>→</b>	<del>)</del>	NVITE				
	ACM	<b>←</b>	+	180 Ringing				
		Ringing tone						
	ANM	<b>←</b>	+	200 OK INVITE				
			-	ACK				
		C	Conversation					
	REL	<b>→</b>	+	BYE				
	RLC	<b>←</b>	+	• 200 OK BYE				

TP610013	SIP refere	nce: RFC 3261 [6]			ISUP reference: 283 027 [1], clause 7.4.21 IU-T Rec. Q.737.1 [33], clause 1.3.5.2.5.2.1
TSS reference:	ISUP-SIP/SS/ UU	S/			
SIP selection criteria:					
ISUP selection criteria:	NOT PICS 11/2				
Test purpose:	Ensure that the SI	JT if the IAM is rece	eived with an <b>ex</b> j	plici	t service 3 request "Not nalling procedure. No response
SIP Parameter	No mapping				
values:					
ISUP Parameter values:					
Comments:	ISUP/BICC		SUT		SIP
	IAM	<b>→</b>		<b>→</b>	INVITE
	ACM	<b>←</b>		<b>←</b>	180 Ringing
		Ringing tone			
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE
				$\rightarrow$	ACK
			Conversation		
	REL	<b>→</b>		<b>→</b>	BYE
	RLC	+		<b>←</b>	200 OK BYE

TP610014	SIP refere	nce: RFC 3261 [6]			ISUP reference: 283 027 [1], clause 7.4.21 "U-T Rec. Q.737.1 [33], clause 1.3.5.2.5.2.1
TSS reference:	ISUP-SIP/SS/ UU	S/			
SIP selection criteria:					
ISUP selection criteria:	NOT PICS 11/2				
Test purpose:	Ensure that the SI		eived with an <b>ex</b> p	olici	t service 3 request "essential" cedure. No response to this
SIP Parameter	No action				
values:					
ISUP Parameter values:					
Comments:	ISUP/BICC		SUT		SIP
	IAM	<b>→</b>		<b>→</b>	INVITE
	ACM	← Ringing tone		<b>←</b>	180 Ringing
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE
				<b>→</b>	ACK
			Conversation		
	REL	<b>→</b>		<b>→</b>	BYE
	RLC	+		<b>←</b>	200 OK BYE

TP610015	SIP refere	nce: RFC 3261 [6]		ISUP reference: 283 027 [1], clause 7.4.21 IU-T Rec. Q.737.1 [33], clause 1.3.5.2.5.2.1
TSS reference:	ISUP-SIP/SS/ UUS	S/		
SIP selection criteria:				
ISUP selection criteria:	NOT PICS 11/1 O	R NOT PICS 11/3		
Test purpose:	no response  Ensure that the St	JT if the FAR is received wi	th an <b>explic</b>	upported in the confirmed state, it service 3 request "Not nalling procedure. No response
SIP Parameter values:	No action			
ISUP Parameter values:				
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
		Ringing tone		
	ANM	<b>←</b>	<b>←</b> →	200 OK INVITE ACK
		Conve	ersation	
	FAR	<b>→</b>		
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	+	<b>←</b>	200 OK BYE

TP610016	SIP reference: RF	FC 3261 [6]		ISUP reference: 283 027 [1], clause 7.4.21 IU-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 unidate component is derived from				
	the User-to-user informati				
SIP Parameter values:	INVITE: User-to-User: uui	data derived from th	e User-to-u	ser information	
ISUP Parameter values:	IAM: User-to-user informa	ation (PIXIT)			
Comments:	ISUP/BICC	SUT		SIP	
	IAM	<b>→</b>	<b>→</b>	INVITE	
	ACM	<b>←</b>	<b>←</b>	180 Ringing	
	Rir	nging tone			
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE	
			→	ACK	
		Conve	rsation		
	REL	<b>→</b>	<b>→</b>	BYE	
	RLC	+	+	200 OK BYE	

TP610017	SIP reference	e: RFC 3261 [6]		ISUP reference: 283 027 [1], clause 7.4.21 IU-T Rec. Q.737.1 [33], clause 1.3.5.2.5.2.1
TSS reference:	ISUP-SIP/SS/ UUS	1		
SIP selection criteria:				
ISUP selection criteria:				
Test purpose:	Ensure that the SUT information paramet from the uuidata cor User-to-User header by the user information the AC	if the 180 Ringing contact is included in the ACM inponent. It starts with the first octetion octets	nins a User-to I and the Us t being the p	o-User header, a User-to-user er-to-user information is derived rotocol discriminator and followed
SIP Parameter values:	180: User-to-User: u	uidata derived from the	User-to-user	information (PIXIT)
ISUP Parameter values:	ACM: User-to-user in	nformation		
Comments:	ISUP/BICC	SUT		SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
		Ringing tone		
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
			ersation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	+	+	200 OK BYE

TP610018	SIP reference: RFC 3	3261 [6]		ISUP reference: 283 027 [1], clause 7.4.21 ГU-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 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 de	erived from the Us	ser-to-user	information (PIXIT)	
ISUP Parameter	ANM: User-to-user informatio	n			
values:					
Comments:	ISUP/BICC	SUT		SIP	
	IAM →		<b>→</b>	INVITE	
	ACM ←		<b>←</b>	180 Ringing	
	Ringin	g tone			
	ANM ←		<del>(</del>	200 OK INVITE	
		_	<b>→</b>	ACK	
	5.5	Conver		D)/5	
	REL →		<b>→</b>	BYE	
	RLC <b>←</b>		+	200 OK BYE	

TP610019	SIP referer	nce: RFC 3261 [6]		ISUP reference: 283 027 [1], clause 7.4.21 TU-T Rec. Q.737.1 [33], clause 1.3.5.2.5.2.1				
TSS reference:	ISUP-SIP/SS/ UUS	SUP-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 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 informa							
SIP Parameter values:	BYE: User-to-User	: uuidata derived from the	User-to-use	r information (PIXIT)				
ISUP Parameter values:	REL: User-to-user	information						
Comments:	ISUP/BICC	SUT		SIP				
	IAM	<b>→</b>	<b>→</b>	INVITE				
	ACM	Pinning tons	<b>←</b>	180 Ringing				
	ANM	Ringing tone	<b>←</b>	200 OK INVITE				
	AINIVI	•	→ →	ACK				
		Conve	ersation	,,,,,,				
	REL	<b>←</b>	<b>←</b>	BYE				
	RLC	<b>→</b>	<b>→</b>	200 OK BYE				

# 6.3.2.11 Explicit call transfer (ECT)

TP611001	SIP reference: RFC	3261 [6]	ES	ISUP reference: 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 FA procedure.	C is received cont	inue without	disrupting the SIP signalling		
SIP Parameter values:	No mapping					
ISUP Parameter values:	LOP: Response "insufficie	nt information"				
Comments:	ISUP/BICC	SUT		SIP		
	IAM	<b>→</b>	<b>→</b>	INVITE		
	ACM	<b>←</b>	<b>←</b>	180 Ringing		
	Ring	ing tone				
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE		
			<b>→</b>	ACK		
		Conve	rsation			
	LOP	<b>→</b>				
	LOP	<b>←</b>				
	FAC(Call transfer active)	<b>→</b>				
			rsation			
	REL	<b>→</b>	<b>→</b>	BYE		
	RLC	<b>+</b>	+	200 OK BYE		

TP611002	SIP reference: RFC	3261 [6]	E	ISUP reference: S 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 I	PICS 13/3		
Test purpose:	Loop prevention procedure supported	not supporte	d, interworking	of "call transfer" indication not
	supported continue without	disrupting the	e SIP signalling	ne loop prevention procedure is not procedure.  ut disrupting the SIP signalling
SIP Parameter	No mapping			
values:	3			
ISUP Parameter				
values:				
Comments:	ISUP/BICC		SUT	SIP
	IAM	<b>→</b>	-	NVITE
	ACM	<b>←</b>	•	180 Ringing
	Ring	ing tone		
	ANM	<b>←</b>	•	200 OK INVITE
			-	► ACK
		C	Conversation	
	LOP	<b>→</b>		
	FAC(Call transfer active)	<b>→</b>		
	,	C	Conversation	
	REL	<b>→</b>	=	<b>▶</b> 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	PICS 12/1 AND PICS 13/3						
criteria:							
Test purpose:	Loop prevention procedure supported	supported, interwo	orking of "call transfer" indication in FAC				
	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 FA set to "sendrecv".	C is received an II	NVITE is sent and the SDP contains an a-line				
SIP Parameter	Re-INVITE SDP a=sendred	:V					
values:							
ISUP Parameter	LOP: Response "insufficie						
values:	FAC: Generic notification	•					
Comments:	ISUP/BICC	SUT	_				
	IAM	<b>→</b>	→ INVITE				
	ACM	<b>.</b>	← 180 Ringing				
		ing tone	# 000 OK INIVITE				
	ANM	~	← 200 OK INVITE → ACK				
		Conve	rsation				
	CPG(hold)	→	→ INVITE(sendonly)				
	or c(noid)	-	← 200 OK INVITE(recvonly)     → ACK				
	LOP	<b>→</b>	-				
	LOP	<b>←</b>					
	FAC(Call transfer, active)	<b>→</b>	<ul><li>→ INVITE(sendrecv)</li><li>← 200 OK INVITE(sendrecv)</li><li>→ ACK</li></ul>				
			rsation				
	REL	<b>→</b>	→ BYE				
	RLC	+	← 200 OK BYE				

TP611004	SIP reference: RFC	3261 [6]	ES	ISUP reference: 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 not supported, interworking of "call transfer" indication in FAC supported  Ensure that the SUT if a LOP(request) is received and the loop prevention procedure is not supported 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	Re-INVITE SDP a=sendred	:V			
values: ISUP Parameter values:	FAC: Generic notification :	= "call transfer, act	ive"		
Comments:	ISUP/BICC	SUT		SIP	
	IAM	<b>→</b>	<b>→</b>	INVITE	
	ACM	<b>←</b>	<b>←</b>	180 Ringing	
	Ring	ing tone			
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE	
			<b>→</b>	ACK	
		Conve	rsation		
	CPG(hold)	<b>→</b>	<b>←</b>	INVITE(sendonly) 200 OK INVITE(recvonly) ACK	
	LOP FAC(Call transfer active)	<b>→</b> →	<b>→</b>	INVITE(sendrecv) 200 OK INVITE(sendrecv) ACK	
			rsation		
	REL RLC	<b>→</b> ←	→ ←	BYE	
	KLC			200 OK BYE	

TP611005	SIP reference: RFC 3261 [6]		ES	ISUP reference: 283 027 [1], clause 7.4.8
TSS reference:	ISUP-SIP/SS/ECT/			
SIP selection				
criteria:				
ISUP selection	PICS 13/3			
criteria:				
Test purpose:	Interworking of "call transfer	r" indication in CP	G supported	
	- 1 1 0UT ( OD			
	Ensure that the SUT if a CP			
SIP Parameter	INVITE is sent and the SDP Re-INVITE SDP a=sendrecy		e sel to sen	diecv.
values:	SDF a=selidlect	V		
ISUP Parameter	CPG: Generic notification =	= "call transfer ac	tive"	
values:	or or concine notineation -	- oan transfor, ao		
Comments:	ISUP/BICC	SUT	1	SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
	ACM	<b>←</b>	<b>←</b>	180 Ringing
	Ringi	ing tone		3 3
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE
			<b>→</b>	ACK
		Conve	ersation	
	CPG(hold)	<b>→</b>	<b>→</b>	INVITE(sendonly)
			<b>←</b>	200 OK INVITE(recvonly)
			<b>→</b>	ACK
	LOP	<b>→</b>		
	CPG(Call transfer active)	<b>→</b>	<b>→</b>	INVITE(sendrecv)
			<b>←</b>	200 OK INVITE(sendrecv)
			<b>→</b>	ACK
		Conve	ersation	
	REL	<b>→</b>	<b>→</b>	BYE
	RLC	+	+	200 OK BYE

#### 6.3.2.12 Anonymous Call Rejection (ACR)

TP612001	ACR-CB Reference: ES 283 027 [1], clause 4.7.1.3.1		Selection criteria:		
TSS reference:	ISUP-SIP/SS/ACR				
Preconditions:					
Test purpose:	Mapping of 433 Anonymit	y Disallowed to REL	cause 24		
				received to due the ACR service	
SIP Parameter	is mapped into a REL cause 24 "call rejected due to ACR supplementary service" 433 Anonymity Disallowed				
values:	1433 Anonymity Disanowet	4			
ISUP Parameter values:	REL cause value 24 "call	rejected due to ACR	? supplemen	tary service"	
Comments:	ISUP	MGCF		SIP	
	IAM	<b>→</b>	<b>→</b>	INVITE	
			<b>←</b>	100 Trying	
	REL(24)	<b>←</b>	<b>←</b>	433 Anonymity Disallowed	
	RLC	<b>→</b>	<b>→</b>	ACK	

TP612002	ACR-CB Reference: ES 283 027 [1], clause 4.7.1.3.1			Selection criteria:
TSS reference:	ISUP-SIP/SS/ACR			
Preconditions:				
Test purpose:	Mapping of 603 Declin	e to REL cause 21		
	Ensure that the 603 De into a REL cause 21 "c		ceived to du	e the ACR service is mapped
SIP Parameter values:	603 Decline			
ISUP Parameter values:	REL cause value 21 "c	call rejected"		
Comments:	ISUP	MGCF		SIP
	IAM	<b>→</b>	<b>→</b>	INVITE
			<b>←</b>	100 Trying
	REL(21)	<b>←</b>	<b>←</b>	603 Decline
	RLC	<b>→</b>	<b>→</b>	ACK

#### 6.3.2.13 Call waiting (CW)

FFS

## 6.3.2.14 Malicious call identification (MCID)

TP614001	MCID Reference: clause 4.7.1.2			Selection criteria: PICS 1/6	
TSS reference:	ISUP-SIP/SS/MCID/				
Preconditions:	ICCI CII /CC/MCID/				
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 requeste	ed in	the sent IDR	
SIP Parameter values:	INFO  XML mcid  request  McidRequestIndicator = "1"				
ISUP Parameter values:	IDR: MCID request indicator: M	IDR: MCID request indicator: MCID requested			
Comments:	ISUP	MGCF		SIP	
	IAM	<b>→</b>	<b>→</b>	INVITE	
			<b>←</b>	100 Trying	
	IDR(MCID request indicator)	<b>←</b>	<b>←</b>	INFO (XML mcid request)	
			<b>→</b>	200 OK INFO	
	ACM	<b>←</b>	<b>←</b>	180 Ringing	
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE ACK	
		Communica	ation		
	REL	<b>→</b>	<b>→</b>	BYE	
	RLC	<b>←</b>	<b>←</b>	200 OK BYE	

TP614002	MCID Reference: clause 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 (	Holding	gIndicator)	
	Ensure that the XML mcid HoldingIndicator is mapped into the MCID request indicator			
	holding requested in the sent ID	DR		
	INFO:			
SIP Parameter	XML mcid			
values:	request			
	HoldingIndicator = "1	"		
ISUP Parameter values:	IDR: Holding indicator (national	<i>l use)</i> :	nolding requeste	ed
Comments:	ISUP		MGCF	SIP
	IAM	<b>→</b>	→	INVITE
			<b>←</b>	100 Trying
	IDR(MCID request indicator)	<b>←</b>	<b>←</b>	INFO (XML mcid request)
			→	200 OK INFO
	ACM	<b>←</b>	<b>←</b>	180 Ringing
	ANINA	<b>←</b>	<b>+</b>	200 OK INWITE
	ANM	~	<b>~</b>	200 OK INVITE ACK
		,	-	ACK
	DEL	7	Communication	BVE
	REL	<b>→</b>	_	BYE
	RLC	+	<u> </u>	200 OK BYE

TP614003	MCID Reference:			Selection criteria:		
17014003	clause 4.7.1.2			PICS 1/6		
TSS reference:	ISUP-SIP/SS/MCID/					
Preconditions:						
Test purpose:	Mapping of IRS (McidResponseIn	dicator)				
	Ensure that MCID response indicator provided, contained in an IRS is mapped into the					
	XML mcid response McidResponseIndicator.					
	INFO:					
	XML mcid					
	request					
SIP Parameter	McidRequestIndicator =	= "1"				
values:	INFO:					
	XML mcid					
	response					
	McidResponseIndicato					
ISUP Parameter	IDR: MCID request indicator: MCI					
values:	IRS: MCID response indicator: MC	CID provided				
Comments:	ISUP	MGC	F	SIP		
	IAM	<b>→</b>	<b>→</b>	INVITE		
			<b>←</b>	100 Trying		
	IDR(MCID request indicator)	<b>←</b>	<b>←</b>	INFO (XML mcid request)		
			<b>→</b>	200 OK INFO		
	IRS (MCID response indicator)	<b>→</b>	<b>→</b>	INFO (XML mcid response)		
	,		<b>←</b>	200 OK INFO		
	ACM	<b>←</b>	<del>-</del>	180 Ringing		
	100 11119119					
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE		
	7 4141	•	<b>•</b>	ACK		
	Communication					
	REL	<b>→</b>	.iioii →	BYE		
	RLC	<del></del>	<del>-</del>	200 OK BYE		
	INLO			ZUU UN DIE		

TP614004	MCID Reference:			Selection criteria:		
17014004	clause 4.7.1.2			PICS 1/6 AND NOT PICS 1/7		
TSS reference:	ISUP-SIP/SS/MCID/					
Preconditions:						
Test purpose:	Mapping of IRS (HoldingProvidedIng	dicator)				
	Ensure that MCID response indicator holding provided, contained in an IRS is mapped into					
	the XML mcid response HoldingProvidedIndicator.					
	INFO:					
	XML mcid					
	request					
SIP Parameter	HoldingIndicator = "1"					
values:	INFO:					
	XML mcid					
	response					
	HoldingProvidedIndicator = "0"					
ISUP Parameter	IDR: Holding indicator (national use)	):				
values:	IRS: Hold provided indicator (national	al use)				
Comments:	ISUP	MGC	F	SIP		
	IAM	<b>→</b>	<b>→</b>	INVITE		
			<del>(</del>	100 Trying		
	IDR(MCID request indicator)	<b>←</b>	<b>←</b>	INFO (XML mcid request)		
	(,		<b>→</b>	200 OK INFO		
	IRS (no MCID response indicator)		<b>→</b>	INFO (XML mcid response)		
	The (no More response maisator)		<b>É</b>	200 OK INFO		
	ACM	<b>←</b>	÷	180 Ringing		
	AOW	•	•	100 Kinging		
	ANM	<b>←</b>	<b>←</b>	200 OK INVITE		
	/ M VIVI	•	→ —	ACK		
		Communica	_	AON		
	REL	Communica →	_	BYE		
		-	<b>→</b>			
	RLC	+	<u> </u>	200 OK BYE		

TP614005	MCID Reference:		Selection criteria:		
17014005	clause 4.7.1.2		PICS 1/6 AND PICS 1/7		
TSS reference:	ISUP-SIP/SS/MCID/				
Preconditions:					
Test purpose:	Mapping of IRS (HoldingProvidedIndicator)				
	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).				
	INFO:				
	XML mcid				
	request				
SIP Parameter	HoldingIndicator = "1"				
values:	INFO:				
	XML mcid				
	response				
ISUP Parameter	HoldingProvidedIndicator = "1"				
values:	IDR: Holding indicator. holding requested				
Comments:	IRS: Hold provided indicator holding provided  ISUP MGCF SIP				
Comments.	IAM	- <del>}</del>	→ INVITE		
	IAW	7	← 100 Trying		
	IDR(MCID request indicator)	<b>←</b>	► INFO (XML mcid request)		
	IDK(MCID request indicator)	~	→ 200 OK INFO		
	IRS (no MCID response indicator)		→ INFO (XML mcid response)		
	(no word response indicator)		← 200 OK INFO		
	ACM	<b>←</b>			
	ACIVI	~	← 180 Ringing		
	ANM	<b>←</b>	€ 200 OK INVITE		
	AINIVI	•	→ ACK		
		Communication	→ NON		
	REL		→ BYF		
	RLC	<del>-</del>	€ 200 OK BYE		
	NLO	~	₹ ZUU UN DIE		

# Annex A (informative): Bibliography

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# History

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