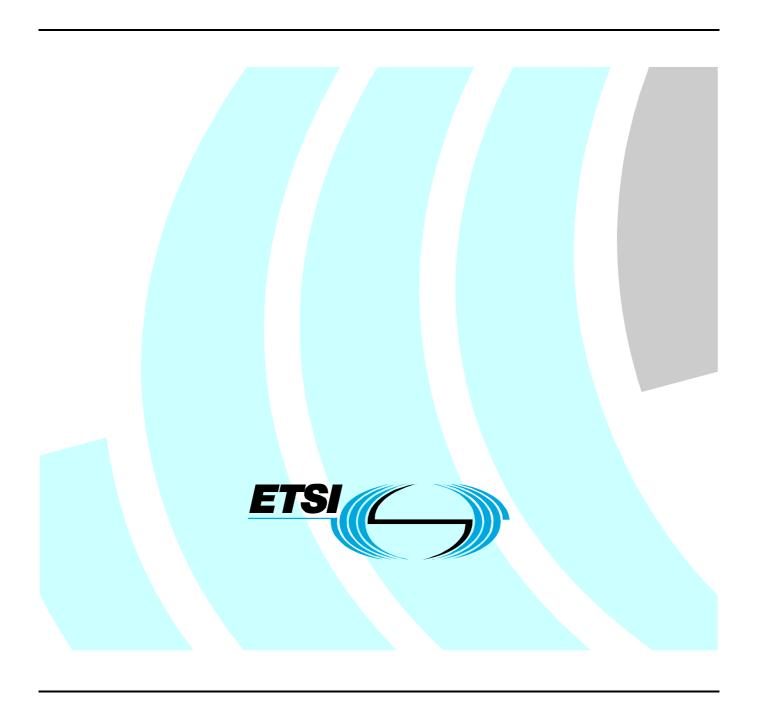
# ETSITS 186 002-3 V1.2.1 (2009-11)

Technical Specification

Telecommunications and Internet Converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol (BICC) or ISDN User Part (ISUP); Part 3: Test Suite Structure and Test Purposes (TSS&TP) for Profile C



#### Reference

RTS/TISPAN-06028-3-NGN-R1

Keywords

BICC, ISUP, SIP, testing, TSS&TP

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#### **Foreword**

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

The present document is part 3 of a multi-part deliverable covering the Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol (BICC) or ISDN User Part (ISUP), as identified below:

- Part 1: "Protocol Implementation Conformance Statement (PICS)";
- Part 2: "Test Suite Structure and Test Purposes (TSS&TP) for Profile A and B";
- Part 3: "Test Suite Structure and Test Purposes (TSS&TP) for Profile C";
- Part 4: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) for Profiles A and B";
- Part 5: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) for Profile C".

### 1 Scope

The present document specifies the network Test Suite Structure and Test Purposes (TSS and TP) Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol (BICCP) ISDN User Part (ISUP) for the Profile C (SIP-I) described in the ITU-T Recommendation Q.1912.5 [1] and EN 383 001 [2].

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 a specific reference, subsequent revisions do not apply.
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#### 2.1 Normative references

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

111	cruding any anici	idinents) appres.
	[1]	ITU-T Recommendation Q.1912.5: "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".
	[2]	ETSI EN 383 001: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control (BICC) Protocol or ISDN User Part (ISUP) [ITU-T Recommendation Q.1912.5, modified]".
	[3]	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".
	[4]	IETF RFC 3261 (2002): "SIP: Session Initiation Protocol".
	[5]	IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".
	[6]	ISO/IEC 9646-1 (1994): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 1: General concepts".
	[7]	ISO/IEC 9646-3 (1992): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 3: The Tree and Tabular Combined Notation (TTCN)".
	[8]	ISO/IEC 9646-7 (1995): "Information technology - Open Systems Interconnection - Conformance

testing methodology and framework - Part 7: Implementation Conformance Statements".

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

#### 2.2 Informative references

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

ITU-T Recommendation Q.730: "ISDN user part supplementary services". [i.1] [i.2] ITU-T Recommendation Q.731: "Stage 3 description for the number identification supplementary services using SS No.7". ITU-T Recommendation Q.731.7: "Malicious call identification (MCID)". [i.3] ITU-T Recommendation Q.732: "Call diversion services". [i.4] ITU-T Recommendation Q.732.7: "Explicit Call Transfer". [i.5]ITU-T Recommendation Q.733: "Stage 3 description for call completion supplementary services [i.6] using Signalling System No. 7: Terminal portability (TP)". ITU-T Recommendation Q.734: "Stage 3 description for multiparty supplementary services using [i.7]Signalling System No. 7: Conference calling". ITU-T Recommendation Q.734.2: "Three-party service". [i.8] [i.9] ITU-T Recommendation Q.735: "Closed user group (CUG)". ITU-T Recommendation Q.737: "User-to-user signalling (UUS)". [i.10] ITU-T Recommendation Q.784: "ISUP basic call test specification". [i.11] ITU-T Recommendations Q.764: "Signalling System No. 7 - ISDN User Part signalling [i.12]procedures".

### 3 Definitions and abbreviations

#### 3.1 Definitions

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

- terms defined in SIP / ISUP interworking reference specification;
- terms defined in ISDN layer 3 reference specification;
- terms defined in ISDN User Part (ISUP) reference specification terms defined in ISO/IEC 9646-1 [6], ISO/IEC 9646-3 [7] and in ISO/IEC 9646-7 [8].

**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 (SUT):** 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 [9]

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

# 3.1.1 SIP Profile C for interworking between SIP with MIME encoding of ISUP and BICC/ISUP

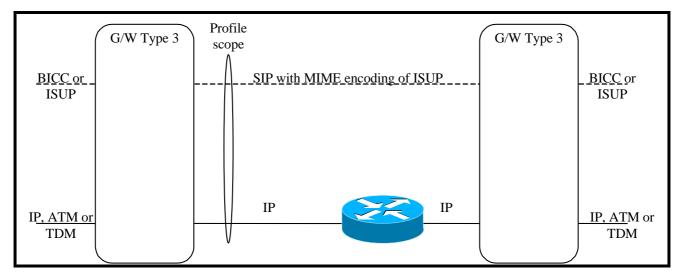


Figure 1: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

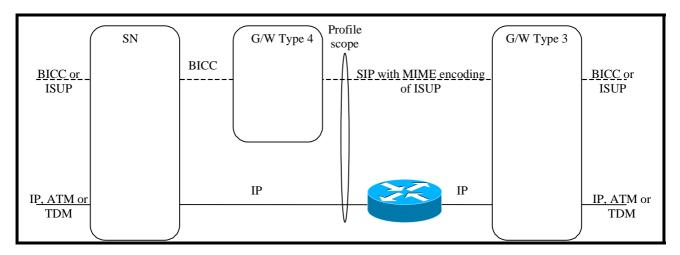


Figure 2: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 and 4 gateways

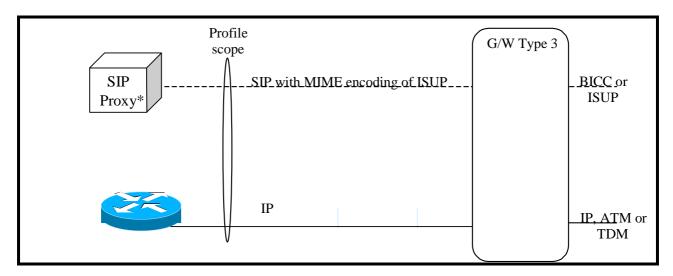


Figure 3: Profile scope for SIP with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateways

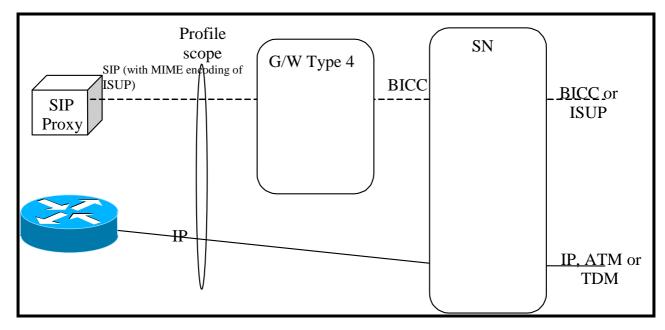


Figure 4: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 4 gateway

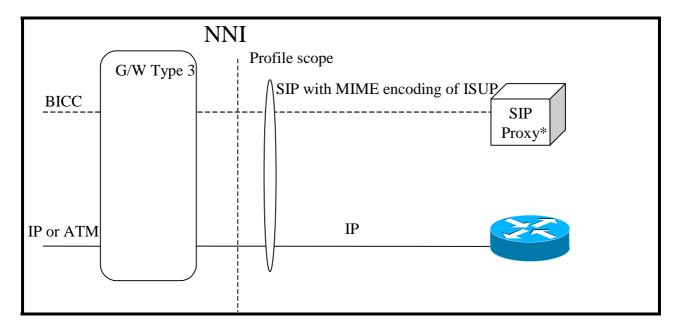


Figure 5: Profile scope for SIP, with MIME encoding of ISUP interworking with BICC/ISUP with type 3 gateway

### 3.2 Abbreviations

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

3PTY	Three-Party
ACM	Address Complete Message
ANM	ANswer Message
ASP	Abstract Service Primitive
ATC	Abstract Test Case
ATM	Abstract Test Method
ATP	Access Transport Parameter
ATS	Abstract Test Suite
AVP	Attribute-Value Pairs

BC Bearer Capability

BCI Backward Call Indicators

BICC Bearer Independent Call Control protocol
BICCP Bearer Independent Call Control Protocol
BLA BLocking Acknowledgement message

BLO BLOcking message CC Country Code

CCBS Completion of Communication to Busy Subscriber

CD Call Deflection
CDIV Call DIVersion
CFB Call Forwarding Busy
CFN ConFusioN message

CFNR Communications Forwarding No Reply

CFU Call Forwarding Unconditional

CGB Circuit Group Blocking

CGBA Circuit Group Blocking Acknowledgement message

CGU Circuit Group Unblocking message

CGUA Circuit Group Unblocking Acknowledgement message

CLIP Calling Line Identification Presentation
CLIR Calling Line Identification Restriction

COL COnnected Line

COLP COnnected Line identification Presentation COLR COnnected Line identification Restriction

CON CONnect message **CONF** CONFerence calling COnTinuity message COT **CPG** Call Progress Message **CPS** Calling Party's Category **CTNb** ConnecTed Number **CUG** Closed User Group Call Waiting CW

DISC DISConnect message
DLE Destination Local Exchange
DSS1 Digital Subscriber System no. 1

ECT Explicit Call transfer
FAA FAcility Accepted message
FAC FACility message

FAR FAcility Request message FCI Forward Call Indicators FRJ Facility ReJect message

GRA circuit Group Reset Acknowledgement message

GRS Group ReSet

HLC High Layer Compatibility

HOLD
IA
Incomming Access
IAM
Initial Address Message
ICB
Incomming Call Barred
IDR
IDentification Request message
I-IWU
Incoming InterWorking Unit

I-MGCF Incoming Media Gateway Control Function

IRS Identification ResponSe message
ISDN Integrated Services Digital Network

ISUP ISDN User Part

ITU International Telecommunication Union

IUTImplementation Under TestLOPLOop Prevention messageMCIDMalicious Call IDentificationMGCFMedia Gateway Control FunctionMIMEMulti-purpose Internet Mail Extension

MOT Means Of Testing

NCI Nature of Connection Indicators NDC National Destination Code OA Outgoing Access

OBCI Optional Backward Call Indicators
O-IWU Outgoing InterWorking Unit
OLE Originating Local Exchange

O-MGCF Outgoing Media Gateway Control Function

OSI Open Systems Interconnection
PCMA Pulse Code Modulation A-law
PCMU Pulse Code Modulation µ-law
PCO Point of Control and Observation

PICS Protocol Implementation Conformance Statement
PIXIT Protocol Implementation eXtra Information for Testing

PT Pay load Type

PTC Parallel Test Component REL RELease message

RES RESUME

RLC ReLease Complete message

RSC ReSet Circuit RTP Real Time Protocol

SAM Subsequent Address Message SDP Session Description Protocol SGM SeGmentation Message SIP Session Initiation Protocol

SIP-I Session Initiation Protocol with encapsulated ISUP

SN Subscriber Number
SS Supplementary Services

SUB SUBaddressing
SUS SUSPEND
SUT System Under Test

TMR Transmission Medium Requirement

TON Type Of Number
TP Test Purpose
TSS Test Suite Structure
UNI User-Network Interface
UPA User Part Available message
UPT User Part Test message
URI Uniform Resource Identifier

USI User Service Information parameter

USR User-to User message UUS User to User Signalling

# 4 Test Suite Structure (TSS)

# 4.1 Interworking from SIP to BICC/ISUP (outgoing call)

SIP -ISUP basic call		
	Sending of the Initial Address Message (IAM)	TP101xxx
	Sending of the Subsequent Address Message (SAM)	TP102xxx
	Sending of COT	TP103xxx
	Receipt of the Address Complete Message (ACM)	TP104xxx
	Receipt of the Call Progress Message (CPG)	TP105xxx
	Receipt of the ANswer Message (ANM)	TP106xxx
	Receipt of the CONnect message (CON)	TP107xxx
	Receipt of the RELease message (REL)	TP108xxx
	Autonomous release at I-IWU	TP109xxx
	Receipt of the BYE, CANCEL message / sending of a REL message	TP110xxx
	Receipt of ReSet Circuit message (RSC), circuit Group ReSet message (GRS) or Circuit Group Blocking message (CGB) with the indication hardware failure oriented	TP111xxx
	Receipt of the SUSPEND Message (SUS)	TP112xxx
	Receipt of the RESUME Message (RES)	TP113xxx

## 4.2 Interworking from BICC/ISUP to SIP (incoming call)

ISUP-SIP basic call		
	Sending of the INVITE message	TP301xxx
	Receipt of the Subsequent Address Message (SAM)	TP302xxx
	Sending of the Address Complete Message (ACM)	TP303xxx
	Sending of the Call Progress Message (CPG)	TP304xxx
	Sending of the ANswer Message (ANM)	TP305xxx
	Sending of the CONnect message (CON)	TP306xxx
	Receipt of the RELease message (REL)	TP307xxx
	Sending of the RELease Message (REL)	TP308xxx
	Receipt of ReSet Circuit message (RSC), circuit Group ReSet	TP309xxx
	message (GRS) or Circuit Group Blocking message (CGB)	
	with the indication hardware failure oriented	
	Receipt of Confusion message	TP310xxx
	Receipt of Suspend message	TP311xxx
	Receipt of a Blocking message	TP312xxx
	Receipt of a user part test message	TP313xxx
	Segmentation	TP314xxx

### 4.3 Supplementary services supported by encapsulation

ISUP-SIP/SIP-ISUP		
	Calling Line Identification Presentation (CLIP)	TP401xxx
	Calling line Identification Restriction (CLIR)	TP402xxx
	COnnected Line identification Presentation (COLP)	TP403xxx
	COnnected Line identification Restriction (COLR)	TP404xxx
	Terminal Portability (TP)	TP405xxx
	SUBaddressing (SUB)	TP406xxx
	Malicious Call IDentification (MCID)	TP407xxx
	Call HOLD (HOLD)	TP408xxx
	Call Waiting (CW)	TP409xxx
	Call DIVersion (CDIV)	TP410xxx
	CONFerence calling (CONF)	TP411xxx
	Explicit Call transfer (ECT)	TP412xxx
	Three-Party (3PTY)	TP413xxx
	User to User Signalling (UUS)	
	User-to-user service 1	TP4140xx
	User-to-user service 2	TP4141xx
	User-to-user service 3	TP4142xx

# 4.4 Interworking SIP-I/ISDN basic call (outgoing)

SIP-I_ISDN basic call outgoing		
	Sending of the SETUP Message	TP501xxx
	Sending of the INFO	TP502xxx
	Receipt of the ALERTING - CALL PROCEEDING -	TP503xxx
	PROGRESS Message	
	Receipt of the CONNECT Message	TP504xxx
	Initiation of the release procedure from the ISDN side	TP505xxx
	Receipt of BYE / CANCEL messages	TP506xxx

# 4.5 Interworking SIP-I/ISDN basic call (incoming)

SIP-I_ISDN basic call incoming		
	Sending of the INVITE message	TP601xxx
	Overlap sending	TP602xxx
	Receipt of the ALERTING - CALL PROCEEDING -	TP603xxx
	PROGRESS Message	
	Sending of the CONNECT message	TP604xxx
	Receipt of the Release message (RELEASE)	TP605xxx
	Receipt of a backward BYE, CANCEL Message	TP606xxx
	Autonomous release at the MG	TP607xxx

### 4.6 Interworking SIP-I/ISDN Supplementary Services

SIP-I_ISDN_Supplementary_Services		
	Calling Line Identification Presentation (CLIP)	TP701xxx
	Calling Line Identification Restriction (CLIR)	TP702xxx
	Connected Line Identification Presentation (COLP)	TP703xxx
	Connected Line Identification Restriction (COLR)	TP704xxx
	Terminal Portability (TP)	TP705xxx
	User-to-User Signalling (UUS)	
	User-to-User Signalling Service 1 (UUS1)	TP7060xx
	User-to-User Signalling Service 2 (UUS2)	TP7061xx
	User-to-User Signalling Service 3 (UUS3)	TP7062xx
	Closed User Group (CUG)	TP707xxx
	SUB-addressing (SUB)	TP708xxx
	Malicious Call Identification (MCID)	TP709xxx
	Conference call (CONF)	TP710xxx
	Explicit Call Transfer (ECT)	TP711xxx
	Call Diversion (CFB, CFNR, CFU, CD)	TP712xxx
	Call HOLD (HOLD)	TP713xxx
	Call Waiting (CW)	TP714xxx
	Three Party Service (3PTY)	TP715xxx

# 5 Test Purposes (TP)

#### 5.1 Introduction

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

#### 5.1.1 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.

### 5.1.2 Source of test purpose definition

The Test Purposes (TPs) have been developed based on ITU-T Recommendation Q.1912.5 [1].

#### 5.1.3 Test purpose structure

The Test Purpose (TP) structure is according to the Test Suite Structure (TSS).

# 5.2 Test purposes for the basic cal

### 5.2.1 Interworking from SIP-I to ISUP (outgoing call)

### 5.2.1.1 Sending of the Initial Address Message (IAM)

TP101001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.1.2 (i,1)		
TSS reference	SIP-ISUP/Basic call/ Sendir	ng of the li	nitial Add	ress Mes	ssage	(IAM)/
SIP selection	NOT PICS 4/4 AND NOT P	ICS 4/5				
criteria						
ISUP selection	NOT PICS 1/6					
criteria						
Test purpose	Ensure that if the SUT upon	receipt o	f the first	INVITE \	with su	ufficient digits, with a SDP
	offer:					
	<ul> <li>the SUT shall delete μ-</li> </ul>	law (PCM	U), if pre	sent, fror	n the i	media description that it will
	send back in the SDP a	answer;				
	<ul> <li>the SUT shall immediat</li> </ul>	tely send o	out the IA	M.		
SIP parameter	SIP INVITE: Audio RTP/AV	P 0 8				
values	200 OK: Audio RTP/AVP 8					
ISUP parameter	IAM USI: A-law or absent					
values						
Comments	SIP-I		SU	T		ISUP
	INVITE(IAM)	→			<b>→</b>	IAM
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
	ACK	→				
	Convers			sation		
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	+			+	RLC

TP101002	SIP reference: RFC 3	3261 [4	1		1	SUP reference:		
		[.,		Q.1	912.5	[1], clause 6.1.2 (i,2ai)		
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5					,		
criteria								
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND F	PICS 4/1					
criteria								
Test purpose		Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP						
	offer 100rel extensions and p							
			/IU), if pre	sent, from	n the i	media description that it will		
	send back in the SDP an	,						
						with the coding of the Nature		
olb .	of Connection Indicators		eter: "CO	T to be ex	(pect	ed".		
SIP parameter	SIP INVITE: Audio RTP/AVP	08						
values	200 OK: Audio RTP/AVP 8							
ISUP parameter	IAM Continuity Indicator: CO			<b>1</b> , USI: A-I	aw or	absent		
values	COT; Continuity Indicator: co	ntinuit		. <del>_</del> 1		LOUID		
Comments	SIP-I		SL	)		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	183 Session Progress	<b>←</b>						
	PRACK	<b>→</b>						
	200 OK PRACK	<b>←</b>				207		
	UPDATE	<b>→</b>			<b>→</b>	COT		
	200 OK UPDATE	+						
	100 D: : (1011)		Preconditi	ons met		1,014		
	180 Ringing(ACM)	+			<u>+</u>	ACM		
	200 OK INVITE(ANM)	<b>←</b>			+	ANM		
	ACK	<b>→</b>						
			Conver	sation				
	BYE(REL)	<b>→</b>			<u>→</u>	REL		
	200 OK BYE(RLC)	<b>←</b>			+	RLC		

TP101003	SIP reference: RFC 3	3261 [4	]			ISUP reference:	
						5 [1], clause 6.1.2 (i,2ai)	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5						
criteria							
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND F	PICS 4/1				
criteria							
Test purpose		Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP					
	offer 100rel extensions and p						
			ЛU), if pre	sent, fro	m the	media description that it will	
	send back in the SDP an	,					
	<ul> <li>the IAM shall be sent out</li> </ul>	immed	liately on	the BIC	C side	with the coding of the Nature	
	of Connection Indicators	parame	eter: "CO	OT to be	expe	cted".	
SIP parameter	SIP INVITE: Audio RTP/AVP	8 0					
values	200 OK: Audio RTP/AVP 8						
ISUP parameter	IAM Continuity Indicator: COT			<b>i</b> , USI: A	\-law o	r absent	
values	COT; Continuity Indicator: co	ntinuit					
Comments	SIP-I		SL	JT		ISUP	
	INVITE(IAM)	<b>→</b>			→	IAM	
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	<b>←</b>					
	UPDATE	<b>→</b>			<b>→</b>	COT	
	200 OK UPDATE	<b>←</b>					
			Preconditi	ions met	t		
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	<b>→</b>					
			Conver	sation			
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC)	+		•	+	RLC	

TP101004	SIP reference: RFC 3	3261 [4]		Q.1		SUP reference: [1], clause 6.1.2 (i,2aii)		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection criteria	PICS 4/4 AND PICS 4/5							
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1							
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Supported header:</li> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer;</li> <li>the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit" or set to "continuity check performed on previous circuit".</li> </ul>							
SIP parameter	SIP INVITE: Audio RTP/AVP							
values	200 OK: Audio RTP/AVP 8							
ISUP parameter						circuit or continuity check		
values	perfe COT Continuity Indicator: cor					SI: A-law or absent		
Comments	SIP-I		SU	Τ		ISUP		
	INVITE(IAM)	<b>→</b>			1	IAM		
	183 Session Progress	<b>→</b>						
	PRACK	+						
	200 OK PRACK	<b>→</b>						
	UPDATE	<b>→</b>			<b>→</b>	COT		
	200 OK UPDATE	+						
		Pre	econdition	ons met				
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	<b>→</b>						
		(	Convers	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101005	SIP reference: RFC	3261 [4]		Q.1		SUP reference: 5 [1], clause 6.1.2 (i,2aii)		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection criteria	PICS 4/4 AND PICS 4/5							
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1							
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Require header:</li> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer;</li> <li>the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit" or set to "continuity check performed on previous circuit".</li> </ul>							
SIP parameter	SIP INVITE: Audio RTP/AVP							
values	200 OK: Audio RTP/AVP 8							
ISUP parameter						circuit or continuity check		
values	perf COT Continuity Indicator: con					SI: A-law or absent		
Comments	SIP-I		SU	T		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	183 Session Progress	<b>→</b>						
	PRACK	+						
	200 OK PRACK	<b>→</b>						
	UPDATE	<b>→</b>			<b>→</b>	СОТ		
	200 OK UPDATE	+						
		Pro	econdition	ons met				
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	<b>→</b>						
			Convers	sation				
	BYE(REL)	<b>→</b>		_	<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101006	SIP reference: RFC 3	3261 [4]			ISUP reference:		
				Q.191	2.5 [1], clause 6.1.2 (i,2b)		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5						
criteria							
ISUP selection	NOT PICS 1/6 AND PICS 4/1						
criteria							
Test purpose	Ensure that if the SUT upon re						
	offer 100rel extensions and p						
			U), if pre	sent, from th	ne media description that it will		
	send back in the SDP and	,					
	<ul> <li>the IAM shall be deferred</li> </ul>		precond	litions have l	peen met.		
SIP parameter	SIP INVITE: Audio RTP/AVP	8 0					
values	200 OK: Audio RTP/AVP 8						
ISUP parameter	IAM USI: A-law or absent						
values		1					
Comments	SIP-I		SU	T	ISUP		
	INVITE(IAM)	<b>→</b>					
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>		-	IAM		
	200 OK UPDATE	+					
			reconditi				
	180 Ringing(ACM)	<b>←</b>		€			
	200 OK INVITE(ANM)	+		€	- ANM		
	ACK	<b>→</b>					
			Convers	sation			
	BYE(REL)	<b>→</b>		-	REL		
	200 OK BYE(RLC)	+		•	- RLC		

TP101007	SIP reference: RFC	SUP reference:						
						5 [1], clause 6.1.2 (i,2b)		
TSS reference	SIP-ISUP/Basic call/ Sendir	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISUP selection	NOT PICS 1/6 AND PICS 4	<del>.</del> /1						
criteria								
Test purpose	offer 100rel extensions and • the SUT shall delete μ-	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Require header:</li> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer;</li> </ul>						
	<ul> <li>the IAM shall be deferred until all preconditions have been met.</li> </ul>							
SIP parameter	SIP INVITE: Audio RTP/AV	P 0 8						
values	200 OK: Audio RTP/AVP 8							
ISUP parameter	IAM USI: A-law or absent							
values			•					
Comments	SIP-I		SU	Т		ISUP		
	INVITE(IAM)	<b>→</b>						
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	+						
	UPDATE	→			<b>→</b>	IAM		
	200 OK UPDATE	+						
		F	Preconditi	ons met				
	180 Ringing(ACM)	<b>←</b>			<del>(</del>	ACM		
	200 OK INVITE(ANM)	+			<del>(</del>	ANM		
	ACK	<b>→</b>						
			Convers	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			<del>(</del>	RLC		

TP101008	SIP reference: RFC 3	3261 [4	SIP reference: RFC 3261 [4] ISUP reference:						
		Q.1912.5 [1], clause 6.1.2 (i,1)							
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	NOT PICS 4/4 AND NOT 4/5					,			
criteria									
ISUP selection criteria	PICS 1/6	PICS 1/6							
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer:</li> <li>the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer;</li> <li>the SUT shall immediately send out the IAM.</li> </ul>								
SIP parameter	SIP INVITE: Audio RTP/AVP	•							
values	200 OK: Audio RTP/AVP 0								
ISUP parameter values	IAM USI: μ-law								
Comments	SIP-I		SU	T		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	<b>→</b>							
			Convers	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP101009	SIP reference: RFC	3261 [4	]	Q.1912	ISUP reference: 2.5 [1], clause 6.1.2 (i,2ai)			
TSS reference	SIP-ISUP/Basic call/ Sendin	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISUP selection	PICS 1/4 AND PICS 1/6 ANI	D PICS	4/1					
criteria								
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Supported header:</li> <li>the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer:</li> <li>the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "COT to be expected".</li> </ul>							
SIP parameter		SIP INVITE: Audio RTP/AVP 0 8						
values	200 OK: Audio RTP/AVP 0							
ISUP parameter	IAM USI: μ-law; Nature of Co	onnectio	n Indicato	rs parameter	: "COT to be expected" COT;			
values	Continuity Indicator: continu	ıity		•	-			
Comments	SIP-I		SU	Т	ISUP			
	INVITE(IAM)	<b>→</b>		→	IAM			
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>		→	СОТ			
	200 OK UPDATE	+						
			Preconditi	ons met				
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		<b>←</b>	ANM			
	ACK	<b>→</b>						
			Convers	sation				
	BYE(REL)	<b>→</b>		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP101010	SIP reference: RFC 3	3261 [4]				reference:	
						, clause 6.1.2 (i,2ai)	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5						
criteria							
ISUP selection	PICS 1/4 AND PICS 1/6 AND	PICS 4	/1				
criteria							
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP						
	offer 100rel extensions and p						
	<ul> <li>the SUT shall delete A-la present in the offer of the</li> </ul>						
	answer;	e media (	uescripii	Jii, iliat it Wii	i Seria i	IL DACK III THE SDF	
	<ul> <li>the IAM shall be sent out</li> </ul>	t immedi	ately on	the BICC sid	le with	the coding of the Nature	
	of Connection Indicators					3	
SIP parameter	SIP INVITE: Audio RTP/AVP			•			
values	200 OK: Audio RTP/AVP 0						
ISUP parameter	IAM USI: μ-law; Nature of Co	nnection	Indicato	rs paramete	r: "CO	T to be expected" COT;	
values	Continuity Indicator: continui	ity					
Comments	SIP-I		SU	IT	ISU	JP	
	INVITE(IAM)	<b>→</b>		-	IAN	Л	
	183 Session Progress	<b>←</b>					
	PRACK	<b>→</b>					
	200 OK PRACK	<b>←</b>					
	UPDATE	<b>→</b>		-	CO	ıΤ	
	200 OK UPDATE	<b>←</b>					
		Р	reconditi	ons met			
	180 Ringing(ACM)	<b>←</b>		•		M	
	200 OK INVITE(ANM)	<b>←</b>		•	- AN	M	
	ACK	<b>→</b>					
			Convers	sation			
	BYE(REL)	<b>→</b>		-			
	200 OK BYE(RLC)	+		€	- RL	С	

TP101011	SIP reference: RFC 3	3261 [4]	Q.		ISUP reference: 5 [1], clause 6.1.2 (i,2aii)				
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection criteria	PICS 4/4 AND PICS 4/5								
ISUP selection criteria	PICS 1/5 AND PICS 1/6 AND PICS 4/1								
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Supported header:</li> <li>the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) ) were present in the offer of the media description, that it will send it back in the SDP answer;</li> <li>the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit" or is set to "continuity check performed on previous circuit".</li> </ul>								
SIP parameter	SIP INVITE: Audio RTP/AVP	0 8							
values	200 OK: Audio RTP/AVP 0								
ISUP parameter values	IAM: USI: μ-law; Continuity check per COT: Continuity Indicator: co	erformed on p	revious circ	cuit	ck required on this circuit"				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	183 Session Progress	+							
	PRACK	<b>→</b>							
	200 OK PRACK	+							
	UPDATE	<b>→</b>		<b>→</b>	COT				
	200 OK UPDATE	+							
			ditions met						
	180 Ringing(ACM)	<del>-</del>		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	<b>→</b>							
			ersation						
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	<b>←</b>		+	RLC				

TP101012	SIP reference: RFC 3		Q.1912.	ISUP reference: 5 [1], clause 6.1.2 (i,2aii)					
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection criteria	PICS 4/4 AND PICS 4/5								
ISUP selection criteria	PICS 1/5 AND PICS 1/6 AND PICS 4/1								
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Require header:</li> <li>the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) ) were present in the offer of the media description, that it will send it back in the SDP answer;</li> <li>the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit" or is set to "continuity check performed on previous circuit".</li> </ul>								
SIP parameter	SIP INVITE: Audio RTP/AVP	0 8							
values	200 OK: Audio RTP/AVP 0								
ISUP parameter values	IAM: USI: μ-law; Continuity ch continuity check perfo COT: Continuity Indicator: cor	rmed on previo	ous circuit	ck required on this circuit"					
Comments	SIP-I		UT	ISUP					
	INVITE(IAM) 183 Session Progress	<b>→</b> ←	<b>→</b>	IAM					
	PRACK 200 OK PRACK	<b>→</b>							
	UPDATE 200 OK UPDATE	<b>→</b> ←	<b>→</b>	СОТ					
		Precondi	tions met						
	180 Ringing(ACM)	+	+	ACM					
	200 OK INVITE(ANM)	+	+	ANM					
	ACK	Conve	rsation						
	BYE(REL)	→ Conve	<b>→</b>	REL					
	200 OK BYE(RLC)	<del>-</del>	<b>+</b>	RLC					

TP101013	SIP reference: RFC	3261 [4	]	0.4	-	SUP reference:		
T00 (	Q.1912.5 [1], clause 6.1.2 (i,2b)							
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISUP selection	PICS 1/6 AND PICS 4/1							
criteria —								
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Supported header:</li> <li>the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer;</li> <li>the IAM shall be deferred until all preconditions have been met.</li> </ul>							
SIP parameter	SIP INVITE: Audio RTP/AVP							
values	200 OK: Audio RTP/AVP 0							
ISUP parameter	IAM USI: μ-law							
values	·							
Comments	SIP-I		SU	T		ISUP		
	INVITE(IAM)	<b>→</b>						
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>			<b>→</b>	IAM		
	200 OK UPDATE	+						
			Preconditi	ons met				
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	<b>→</b>						
			Convers	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101014	SIP reference: RFC 3261 [4]		ISUP reference:				
						5 [1], clause 6.1.2 (i,2b)	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5						
criteria							
ISUP selection	PICS 1/6 AND PICS 4/1						
criteria							
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with a SDP offer 100rel extensions and preconditions extensions in the SIP Require header:</li> <li>the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer;</li> <li>the IAM shall be deferred until all preconditions have been met.</li> </ul>						
SIP parameter	SIP INVITE: Audio RTP/AVP						
values	200 OK: Audio RTP/AVP 0						
ISUP parameter	IAM USI: μ-law						
values							
Comments	SIP-I		SL	IT		ISUP	
	INVITE(IAM)	<b>→</b>					
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>			<b>→</b>	IAM	
	200 OK UPDATE	+					
			Preconditi	ons met			
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	<b>→</b>					
			Conver	sation			
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC)	<b>←</b>			+	RLC	

TP101015	SIP reference: RFC	3261 [4]		Q.191 Q.191	ISUP reference:  2.5 [1], clause 6.1.3.2  2.5 [1], clause 6.1.3.3  2.5 [1], clause 6.1.3.4			
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection criteria								
ISUP selection criteria	NOT PICS 1/9 AND NOT PICS 4/4 and NOT PICS 4/5							
Test purpose	<ul> <li>Ensure that the SUT on receipt of an INVITE message sends an IAM message, where:</li> <li>the Calling party's category is generated from the Calling Party's Category present in the encapsulated IAM;</li> <li>the Nature of Connection Indicators (NCI) is generated by the MGCF using the Nature of Connection Indicators received in the encapsulated IAM;</li> <li>the appropriate values of the Forward Call Indicator parameter are generated by the MGCF using the Forward Call Indicators parameter present within the received encapsulated IAM.</li> </ul>							
SIP parameter values								
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
		C	onversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

P101016	SIP reference: RF	C 3261 [4]			ISUP reference:				
				Q.191	2.5 [1], clause 6.1.3.5				
TSS reference	SIP-ISUP/Basic call/ Sendi	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection criteria	NOT PICS 4/4 and NOT PICS 4/5								
ISUP selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message with an encapsulated IAM message.  The TMR and USI shall be taken from the encapsulated ISUP:  sends an IAM message, with the Transmission Medium Requirement (TMR) taken from the encapsulated ISUP.								
SIP parameter values	SIP INVITE								
ISUP parameter values	IAM; USI; ISDN_BC_ITR; 1	ΓMR							
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		<b>←</b>	ACM				
	200 OK INVITE(ANM)	+		<b>←</b>	ANM				
	ACK	<b>→</b>							
		(	Conversati	on					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

Values and selection criteria for the test purpose TP101020					
VA_01	USI= speech	ISUP_TMR = speech			
VA_02	USI= 3,1 kHz audio	ISUP_TMR = 3,1 kHz audio			
VA_03	USI= unrestricted digital information	ISUP_TMR = 64 kbits/s unrestricted			
	ISDN_BC_ITR = 64 kbits/s unrestricted				
VA_04	No USI contained in the encapsulated IAM	ISUP_TMR = speech			
VA_05	No USI contained in the encapsulated IAM	ISUP_TMR = 3,1 kHz audio			
VA_06	No USI contained in the encapsulated IAM	ISUP_TMR = 64 kbits/s unrestricted			

TP101017	SIP reference: RFC 3261 [4]			-	SUP reference:				
		Q.1912.5 [1], clause 6.1.3.5							
TSS reference	SIP-ISUP/Basic call/ Sendi	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	NOT PICS 4/4 and NOT PI	CS 4/5							
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT in the								
	encapsulated IAM message	e the HLC sha	ll be taken f	rom the	encapsulated ISUP:				
	<ul> <li>sends an IAM messag</li> </ul>								
SIP parameter	INVITE;				·				
values									
ISUP parameter	IAM; Access transport pa	rameter HLC:	HLC_VALU	JE; USI					
values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	<b>→</b>							
		Co	nversation						
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

	Values and selection criteria for the test purpose TP1010017				
VA_01	HLC_VALUE = Telephony				
	USI= speech				
VA_02	HLC_VALUE = Facsimile Group 2/3				
	USI= 3,1 kHz audio				
VA_03	HLC_VALUE == Facsimile Group 4 Class I				
	USI= Unrestricted digital information				
VA_04	HLC_VALUE == Teletex service, basic and mixed mode of operation and facsimile service				
	Group 4, Classes II and III				
	USI= Unrestricted digital information				
VA_05	HLC_VALUE == Teletex service, basic and processable mode of operation				
	USI= Unrestricted digital information				
VA_06	HLC_VALUE = Teletex service, basic mode of operation				
	USI= Unrestricted digital information				
VA_07	HLC_VALUE = Syntax based Videotex				
	USI= Unrestricted digital information				
VA_08	HLC_VALUE = International Videotex interworking via gateways or interworking units				
	USI= Unrestricted digital information				
VA_09	HLC_VALUE = Telex service				
	USI= Unrestricted digital information				
VA_10	HLC_VALUE = Message Handling Systems (MHS)				
	USI= Unrestricted digital information				
VA_11	HLC_VALUE = OSI application				
	USI= Unrestricted digital information				
VA_12	HLC_VALUE = Audio visual				
	USI= Unrestricted digital information				

TP101018	SIP reference: RFC 3261 [4] ISUP reference:							
					2.5 [1], clause 6.1.3.9			
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection criteria	NOT PICS 4/4 and NOT PICS 4/5							
ISUP selection criteria	PICS 4/3							
Test purpose	Ensure that the MGCF acting as an independent exchange and shall perform the normal BICC/ISUP Hop Counter procedure using the Hop Counter taken from the encapsulated IAM if the Hop Counter parameter is available.  The initial and successively mapped values of Hop Counter should be large enough to accommodate the maximum number of hops that might be expected of a validly routed call.							
SIP parameter values	Max-Forwards header							
ISUP parameter values	IAM: Hop Counter parameter	· value						
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
		C	onversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP101019	SIP reference: RFC 32	261 [4]		SUP reference: 2.5 [1], clause 6.1.3.1			
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 1/9 AND NOT PICS 4/4	and NOT PICS	4/5				
criteria							
ISUP selection	NOT PICS 1/7						
criteria							
Test purpose	<ul> <li>Ensure that the SUT on receipt of an INVITE message with a Called party number contained in the user info component of the Request-URI.</li> <li>Send an IAM Message with the called party number coded as follows:</li> <li>Nature of address indicator:</li></ul>						
SIP parameter							
values	1						
ISUP parameter values	IAM: Called party number						
Comments	SIP-I	Sl	JT	ISUP			
	INVITE(IAM)	<b>→</b>	<b>→</b>	IAM			
	180 Ringing(ACM)	<del>(</del>	+	ACM			
	200 OK INVITE(ANM)	<del>-</del>	+	ANM			
	ACK	<b>→</b>					
		Conver					
	BYE(REL)	<b>→</b>	→	REL			
	200 OK BYE(RLC)	<del>-</del>	<del>-</del>	RLC			

TP101020	SIP reference: RF0	C 3261 [4]			ISUP reference:		
					2.5 [1], clause 6.1.3.1		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 1/9 AND NOT PICS 4	4/4 and NOT PIC	CS 4/5				
criteria							
ISUP selection	PICS 1/7						
criteria							
Test purpose	Ensure that the SUT on receipt of an INVITE message with a Called party number contained in the user info component of the Request-URI.  Send an IAM Message with the called party number coded as follows:  Nature of address indicator:						
	Analyse the information contained in received URI with user=phone, and if it is in the format:  +CC NDC SN where CC is not the country code of the network in which the next hop terminates, then set Nature of Address indicator to "International number", remove "+" and use the remaining digits to fill the Address signals.  Internal Network Number Indicator: routing to internal network number not allowed  Numbering plan Indicator 001 ISDN (Telephony) numbering plan  Address Signals CC NDC SN						
SIP parameter							
values							
ISUP parameter	IAM: Called party number						
values	OID I	1	OUT	1	LOUID		
Comments	SIP-I		SUT	_	ISUP		
	INVITE(IAM)	→ ←		<b>→</b>	IAM ACM		
	180 Ringing(ACM) 200 OK INVITE(ANM)	+		+	ANM		
	ACK	<b>→</b>		7	AINIVI		
	ACK		varaation				
	DVE(DEL)		ersation/		DEL		
	BYE(REL)	→ ←		<b>→</b>	REL		
	200 OK BYE(RLC)	<b>—</b>		<b>←</b>	RLC		

TP101021	SIP reference: RFC 3	3261 [4]	ı	SUP reference:				
			EN 383 0	01 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	NOT PICS 4/4 AND NOT PIC	S 4/5 AND PICS	S 1/9					
criteria								
ISUP selection criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a-Law, then independent from the received order of preference:  the G.711 a-law codec shall be returned in the SDP answer as preferred codec.							
SIP parameter	Offer: m=audio 4711 RTF	P/AVP 0 8						
values	Answer: m=audio 4712 RTF	P/AVP 8 0						
ISUP parameter values								
Comments	SIP-I	SU	JT	ISUP				
	INVITE(IAM)	<b>→</b>	<b>→</b>	IAM				
	180 Ringing(ACM)	<b>←</b>	+	ACM				
	200 OK INVITE(ANM)	<b>←</b>	<b>←</b>	ANM				
	ACK	<b>→</b>						
		Conver	sation					
	BYE(REL)	<b>→</b>	→	REL				
	200 OK BYE(RLC)	+	+	RLC				

TP101022	SIP reference: RFC	3261 [4]			ISUP reference:		
				EN 383 0	01 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 ANI	D PICS 1/9					
criteria							
ISUP selection	PICS 1/4 AND NOT PICS 1/	6 AND PICS 4	/1				
criteria							
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a-						
		Law 100rel extensions and preconditions extensions in the SIP Supported header, then					
	independent from the received order of preference:						
					with the coding of the Nature		
	of Connection Indicators	•		•			
	• the G.711 a-law codec shall be returned in the SDP answer as preferred codec.						
SIP parameter	Offer: m=audio 4711 RT						
values	Answer: m=audio 4712 RT						
ISUP parameter	IAM: Continuity Indicator: CC		cted, U	SI: A-law c	or absent		
values	COT: Continuity Indicator: co	ontinuity			T		
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		→	IAM		
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	<b>←</b>					
	UPDATE	<b>→</b>		→	СОТ		
	200 OK UPDATE	<del>-</del>					
	180 Ringing(ACM)	<del>-</del>		+	ACM		
	200 OK INVITE(ANM)	<del>-</del>		+	ANM		
	ACK	→					
		Con	versatio	n			
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	<b>←</b>		<b>←</b>	RLC		

TP101023	SIP reference: RFC 3	3261 [4]		ISUP reference:			
			EN 383 0	01 [2], clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1/9					
criteria							
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND PICS 4/1					
criteria							
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a- Law 100rel extensions and preconditions extensions in the SIP Require header, <b>then</b> <b>independent from the received order of preference:</b>						
	<ul> <li>the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "COT to be expected";</li> <li>the G.711 a-law codec shall be returned in the SDP answer as preferred codec.</li> </ul>						
SIP parameter	Offer: m=audio 4711 RTF	P/AVP 0 8					
values	Answer: m=audio 4712 RTF	P/AVP 8 0					
ISUP parameter	IAM: Continuity Indicator: CO		ed, USI: A-law o	or absent			
values	COT: Continuity Indicator: co	ntinuity					
Comments	SIP-I	SI	JT	ISUP			
	INVITE(IAM)	<b>→</b>	→	IAM			
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>	→	COT			
	200 OK UPDATE	+					
	180 Ringing(ACM)	+	+	ACM			
	200 OK INVITE(ANM)	+	+	ANM			
	ACK	<b>→</b>					
		Conve	rsation				
	BYE(REL)	<b>→</b>	→	REL			
	200 OK BYE(RLC)	+	+	RLC			

TP101024	SIP reference: RFC 3	3261 [4]		ISUP reference:				
				3 001 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9							
criteria								
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1							
criteria								
Test purpose				h a SDP offer for μ-Law and a-				
				e SIP Supported header:, <b>then</b>				
	independent from the recei							
				de with the Continuity check				
	indicator "continuity che		n this circuit	or continuity cneck				
	performed on previous		lia tha CDD a	nower as professed and a				
SIP parameter	the G.711 a-law codec shall be returned in the SDP answer as preferred codec.  Offer: m=audio 4711 RTP/AVP 0 8							
values	Answer: m=audio 4711 RTF							
ISUP parameter			required on t	this circuit or continuity check				
values	performed on previo							
Values	COT: Continuity Indicator: co			ont				
Comments	SIP-I		SUT	ISUP				
	INVITE(IAM)	<b>→</b>	1 -	<b>▶</b> IAM				
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>		COT				
	200 OK UPDATE	+						
	180 Ringing(ACM)	+	•	ACM				
	200 OK INVITE(ANM)	+	•	ANM ANM				
	ACK	<b>→</b>						
			ersation					
	BYE(REL)	<b>→</b>		<b>→</b> REL				
	200 OK BYE(RLC)	<del>-</del>	( €	RLC				

TP101025	SIP reference: RFC 3	3261 [4]	1	ISUP reference:				
				01 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9							
criteria								
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1							
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a-							
	Law 100rel extensions and pr			SIP Require header, <b>then</b>				
	independent from the receive							
				with the Continuity check				
		ck required or	n this circuit"" (	continuity check performed				
	on previous circuit";							
	the G.711 a-law codec sh		in the SDP ans	wer as preferred codec.				
SIP parameter	Offer: m=audio 4711 RTF							
values	Answer: m=audio 4712 RTF	,,,,,						
ISUP parameter				s circuit or continuity check				
values	performed on previou			t				
Comments	COT: Continuity Indicator: co			LICLUD				
Comments	•		UT	ISUP				
	INVITE(IAM)	<b>→</b>	<b>→</b>	IAM				
	183 Session Progress	<del>(</del>						
	PRACK	<b>→</b>						
	200 OK PRACK	+		0.07				
	UPDATE	<b>→</b>	<b>→</b>	COT				
	200 OK UPDATE	<del>(</del>						
	180 Ringing(ACM)	<del>-</del>	<del>-</del>	ACM				
	200 OK INVITE(ANM)	<b>←</b>	<b>←</b>	ANM				
	ACK	<b>→</b>						
			rsation					
	BYE(REL)	<b>→</b>	<b>→</b>	REL				
	200 OK BYE(RLC)							

TP101026	SIP reference: RFC 3261 [4]			ISUP reference:				
	EN 383 001 [2], clause 6.1.3.5.							
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9							
criteria								
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND NOT PICS 4/1							
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a-							
						IP Supported header:, then		
	independent from the rece							
	<ul> <li>the shall be deferred un</li> </ul>							
	the G.711 a-law codec s			n the SDI	o ans	wer as preferred codec.		
SIP parameter	Offer: m=audio 4711 RTP/AVP 0 8							
values	Answer: m=audio 4712 R	ΓP/AVP (	8 0					
ISUP parameter								
values								
Comments	SIP-I		SL	JT		ISUP		
	INVITE(IAM)	<b>→</b>						
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	→			<b>→</b>	IAM		
	200 OK UPDATE	+						
	180 Ringing(ACM)	+			<b>←</b>	ACM		
	200 OK INVITE(ANM)	<b>←</b>			<b>←</b>	ANM		
	ACK	<b>→</b>						
			Conver	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101027	SIP reference: RFC 3261 [4]			ISUP reference:					
				EN 383 0	01 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 4/4 AND PICS 4/5 AND PICS 1/9							
criteria									
ISUP selection	PICS 1/5 AND NOT PICS 1/6	6 and not	PICS 4/	′1					
criteria									
Test purpose		Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a-							
	Law 100rel extensions and p				IP Require header, then				
	independent from the rece								
	<ul> <li>the shall be deferred unit</li> </ul>								
	<ul> <li>the G.711 a-law codec s</li> </ul>		rned in t	he SDP ans	wer as preferred codec.				
SIP parameter	Offer: m=audio 4711 RTP/AVP 0 8								
values	Answer: m=audio 4712 RT	P/AVP 8 0							
ISUP parameter									
values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>							
	183 Session Progress	<b>←</b>							
	PRACK	<b>→</b>							
	200 OK PRACK	<b>←</b>							
	UPDATE	→		→	IAM				
	200 OK UPDATE	<b>←</b>							
	180 Ringing(ACM)	<b>←</b>		<del>(</del>	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	<b>→</b>							
		С	onversat	ion					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP101028	SIP reference: RFC 3261 [4]			ISUP reference: EN 383 001 [2], clause 6.1.3.5.2.2					
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	NOT PICS 4/4 AND NOT F	NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9							
criteria									
ISUP selection criteria	PICS 1/7								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law, then independent the normal offer answer procedures apply:  • the G.711 a-law codec shall be returned in the SDP answer.								
SIP parameter	Offer: m=audio 4711 R	Offer: m=audio 4711 RTP/AVP 8							
values	Answer: m=audio 4711 R	Answer: m=audio 4711 RTP/AVP 8							
ISUP parameter values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		<del>-</del>	ACM				
	200 OK INVITE(ANM)								
	ACK	<b>→</b>							
		Conversation							
	BYE(REL)	→		<b>→</b>	REL				
	200 OK BYE(RLC) ← RLC								

TP101029	SIP reference: RFC 3261 [4]				-	SUP reference:		
						01 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9							
criteria								
ISUP selection	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1							
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no							
						SIP Supported header:, then		
	independent the normal off							
						with the coding of the Nature		
	of Connection Indicators							
	<ul> <li>the G.711 a-law codec sl</li> </ul>	hall be retu	ırned i	n the SDF	ansv	wer.		
SIP parameter	Offer: m=audio 4711 RTP/AVP 8							
values	Answer: m=audio 4711 RTF							
ISUP parameter	IAM: Continuity Indicator: CO		pecte	<b>d</b> , USI: A-	law o	r absent		
values	COT: Continuity Indicator: co	ntinuity				_		
Comments	SIP-I		SU	Т		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>			<b>→</b>	COT		
	200 OK UPDATE	+						
		Pre	conditi	ons met				
	180 Ringing(ACM)	+			<b>←</b>	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	ACK	<b>→</b>						
		С	onvers	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP101030	SIP reference: RFC 3261 [4]		ISUP reference:						
						01 [2], clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9								
criteria									
ISUP selection	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1								
criteria									
Test purpose	Ensure that the SUT on receip								
	μ-Law 100rel extensions and					SIP Require header, then			
	independent the normal offe								
						with the coding of the Nature			
	of Connection Indicators								
oin .	the G.711 a-law codec shall be returned in the SDP answer.								
SIP parameter	Offer: m=audio 4711 RTP/AVP 8								
values	Answer: m=audio 4711 RTP/AVP 8  IAM: Continuity Indicator: COT to be expected, USI: A-law or absent								
ISUP parameter			expecte	<b>a</b> , USI: A	-law o	or absent			
values	COT: Continuity Indicator: co	ntinuity	011	_		LOUID			
Comments	SIP-I		SU			ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress	<b>←</b>							
	PRACK								
	200 OK PRACK UPDATE	<b>←</b>			<b>→</b>	COT			
	· · · · ·				7	COT			
	200 OK UPDATE ←								
	Preconditions met								
	180 Ringing(ACM)	<del>+</del>			+	ACM			
	200 OK INVITE(ANM)	<b>→</b>			~	ANM			
	ACK	7	C						
	DVE(DEL)	_	Convers	sation	_	DEL			
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			-	RLC			

TP101031	SIP reference: RFC 3261 [4]					SUP reference:			
						01 [2], clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9								
criteria									
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1								
criteria									
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no μ-Law 100rel extensions and preconditions extensions in the SIP Supported header:, <b>the</b>								
	independent the normal off								
						with the Continuity check			
	indicator "continuity che			inis circi	lit" OI	"continuity check			
	performed on previous			41 ODI					
OID	• the G.711 a-law codec shall be returned in the SDP answer.								
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8								
	Answer: m=audio 4711 RT				4la!	- circuit or continuity chools			
ISUP parameter values	performed on previo					s circuit or continuity check			
values	COT: Continuity Indicator: co					L			
Comments	SIP-I		SU			ISUP			
	INVITE(IAM)	→			<b>→</b>	IAM			
	183 Session Progress	+							
	PRACK	<b>→</b>							
	200 OK PRACK	+							
	UPDATE	<b>→</b>			<b>→</b>	СОТ			
	200 OK UPDATE	+							
		Р	reconditi	ons met					
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	<b>→</b>							
			Convers	sation					
	BYE(REL)	<b>→</b>			<b>↑</b>	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP101032	SIP reference: RFC 3261 [4]					SUP reference:			
						01 [2], clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9								
criteria									
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6	AND PIC	S 4/1						
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no µ-Law 100rel extensions and preconditions extensions in the SIP Require header, <b>then</b>								
	independent the normal offer answer procedures apply:								
						with the Continuity check			
	indicator "continuity che								
	performed on previous		eu on	ins circ	uit O	continuity check			
	<ul> <li>the G.711 a-law codec sh</li> </ul>		urned ir	the SDI	D ane	MAr.			
SIP parameter	Offer: m=audio 4711 RTF		unica ii	T the ODI	ans	wei.			
values	Answer: m=audio 4711 RTF								
ISUP parameter		-	heck re	auired c	n thi	s circuit or continuity check			
values	performed on previous								
	COT: Continuity Indicator: co								
Comments	SIP-I		SU	Т		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress	+							
	PRACK	<b>→</b>							
	200 OK PRACK	+							
	UPDATE	<b>→</b>			<b>→</b>	COT			
	200 OK UPDATE	+							
		Pre	econdition	ons met					
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	<b>→</b>							
			Convers	sation					
	BYE(REL)	→			<b>→</b>	REL			
	200 OK BYE(RLC)	<b>←</b>			<b>←</b>	RLC			

TP101033	SIP reference: RFC	3261 [4]				ISUP reference:	
				EN	1 383 O	01 [2], clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AN	D PICS 1/9					
criteria							
ISUP selection	NOT PICS 1/6 AND NOT PI	ICS 4/1					
criteria							
Test purpose						a SDP offer for a-Law and no	
						SIP Supported header:, then	
	independent the normal of						
	the IAM shall be deferred.						
	the G.711 a-law codec		ırned i	n the SL	OP ans	wer.	
SIP parameter	Offer: m=audio 4711 R						
values	Answer: m=audio 4711 R	IP/AVP 8					
ISUP parameter							
values	OID I		01			TIQUE	
Comments	SIP-I	$+$ $\leftarrow$	SL	)		ISUP	
	INVITE(IAM)	<b>→</b>					
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+				1000	
	UPDATE	<b>→</b>			<b>→</b>	IAM	
	200 OK UPDATE	+					
	180 Ringing(ACM)	+			<del>-</del>	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	<b>→</b>					
			conver	sation			
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC)	+			<b>←</b>	RLC	

TP101034	SIP reference: RFC	3261 [4]			SUP reference:		
				EN 383 0	01 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1/9	)				
criteria							
ISUP selection	NOT PICS 1/6 AND NOT PIC	CS 4/1					
criteria							
Test purpose					SDP offer for a-Law and no		
	μ-Law 100rel extensions and				SIP Require header, then		
	independent the normal off						
	the IAM shall be deferred						
	<ul> <li>the G.711 a-law codec s</li> </ul>		urned in	the SDP ans	wer.		
SIP parameter	Offer: m=audio 4711 RT	-					
values	Answer: m=audio 4711 RT	P/AVP 8					
ISUP parameter							
values							
Comments	SIP-I		SU	Γ	ISUP		
	INVITE(IAM)	→					
	183 Session Progress	+					
	PRACK	→					
	200 OK PRACK	+					
	UPDATE	→		<b>→</b>	IAM		
	200 OK UPDATE	+					
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	<b>→</b>					
		(	Convers	ation			
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP101035	SIP reference: RFC 3	261 [4]		SUP reference:						
		EN 383 001 [2], clause 6.1.3.5.2.2								
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	NOT PICS 4/4 AND NOT PIC	S 4/5 AND PIC	S 1/9							
criteria										
ISUP selection	PICS 1/7									
criteria	Francis that the CLIT on reasin	-4 -4 INI\/ITC		CDD affair in line with aut						
Test purpose	Ensure that the SUT on receip	of an invite	message with a	SDP offer m line without						
	a-law codec:									
	the u-law codec shall be	•								
SIP parameter	Offer: m=audio 4711 RTP	-								
values	m=audio 4712 RTP	-								
	Answer: m=audio 0 RTP/AV	'P 0								
ISUP parameter										
values										
Comments	SIP-I	SI	JT	ISUP						
	INVITE(IAM)	<b>→</b>	→	IAM						
	180 Ringing(ACM)	+	+	ACM						
	200 OK INVITE(ANM)	+	+	ANM						
	ACK	ACK →								
		Conversation								
	BYE(REL)	<b>→</b>	<b>→</b>	REL						
	200 OK BYE(RLC)	+	+	RLC						

TP101036	SIP reference: RFC 3	3261 [4]				SUP reference:		
				EN	383 0	01 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 4/4 AND PICS 4/5 AND PICS 1/9						
criteria								
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND P	ICS 4/1					
criteria								
Test purpose	Ensure that the SUT on recei							
						in the SIP Supported header:		
						with the coding of the Nature		
	of Connection Indicators			Γto be e	xpect	ed";		
	the u-law codec shall be							
SIP parameter	Offer: m=audio 4711 RTF							
values	m=audio 4712 RTF		3					
10115	Answer: m=audio 0 RTP/A\							
ISUP parameter	IAM: Continuity Indicator: CO			d, USI: A	-law c	or absent		
values	COT: Continuity Indicator: co	ntinuity			1	lious		
Comments	SIP-I		SU			ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	183 Session Progress	<del>(</del>						
	PRACK	<b>→</b>						
	200 OK PRACK	<del>(</del>				207		
	UPDATE	<b>→</b>			<b>→</b>	СОТ		
	200 OK UPDATE	+						
	100 5: (4011)	L .	Preconditi	ons met		1.014		
	180 Ringing(ACM)	<del>(</del>			+	ACM		
	200 OK INVITE(ANM)	<del>(</del>			+	ANM		
	ACK	→						
	5)(5(551)	<b> </b>	Conver	sation		55		
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	<b>←</b>			<b>←</b>	RLC		

TP101037	SIP reference: RFC 3	3261 [4]			_	SUP reference:	
						01 [2], clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1	/9				
criteria							
ISUP selection	PICS 1/4 AND NOT PICS 1/6	AND P	ICS 4/1				
criteria							
Test purpose	Ensure that the SUT on recei						
	a-law codec 100rel extension						
						with the coding of the Nature	
	of Connection Indicators			to be ex	<b>kpect</b>	ed";	
	<ul> <li>the u-law codec shall b</li> </ul>						
SIP parameter	Offer: m=audio 4711 RTF	-					
values	m=audio 4712 RTF						
	Answer: m=audio 0 RTP/A\						
ISUP parameter	IAM: Continuity Indicator: CO			d, USI: A	-law o	r absent	
values	COT: Continuity Indicator: co	ntinuity					
Comments	SIP-I		SU	Т		ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>			<b>→</b>	COT	
	200 OK UPDATE	+					
			reconditi	ons met			
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
	ACK	<b>→</b>					
			Convers	sation			
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC)	<b>←</b>			+	RLC	

TP101038	SIP reference: RFC 3	3261 [4]		<b>EN 0</b>		SUP reference:	
						11 [2], clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1/9	9				
criteria							
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6	AND PIC	CS 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer m line without  a-law codec 100rel extensions and preconditions extensions in the SIP Supported header:  the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit" or "continuity check performed on previous circuit";						
	<ul> <li>the u-law codec shall b</li> </ul>	e rejecte	d.				
SIP parameter	Offer: m=audio 4711 RTF	P/AVP 0					
values	m=audio 4712 RTF						
	Answer: m=audio 0 RTP/A\	/P 0					
ISUP parameter	IAM: Continuity Indicator: continuity check required on this circuit or continuity check						
values	performed on previo COT: Continuity Indicator: co						
Comments	SIP-I		SU	Τ		ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	183 Session Progress	+					
	PRACK	<b>→</b>					
	200 OK PRACK	+					
	UPDATE	<b>→</b>			<b>→</b>	СОТ	
	200 OK UPDATE	+					
		Pro	econditio	ons met			
	180 Ringing(ACM)	+			<del>(</del>	ACM	
	200 OK INVITE(ANM)	+			<del>(</del>	ANM	
	ACK	<b>→</b>					
			Convers	ation			
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC)	+			<del>(</del>	RLC	

TP101039	SIP reference: RFC 3	3261 [4]				SUP reference:		
				EN 3	83 0	01 [2], clause 6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection		PICS 4/4 AND PICS 4/5 AND PICS 1/9						
criteria								
ISUP selection	PICS 1/5 AND NOT PICS 1/6	AND P	ICS 4/1					
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer m line without							
		<ul> <li>a-law codec 100rel extensions and preconditions extensions in the SIP Require header:</li> <li>the IAM shall be sent out immediately on the ISUP side with the Continuity check</li> </ul>						
	indicator "continuity che			this circu	I <b>It</b> " OI	"continuity check		
	performed on previous							
CID managed an	the u-law codec shall be							
SIP parameter values	Offer: m=audio 4711 RTF m=audio 4712 RTF							
values	Answer: m=audio 0 RTP/A\	-						
ISUP parameter			chack ro	auired or	a thi	s circuit or continuity check		
values		IAM: Continuity Indicator: continuity check required on this circuit or continuity check performed on previous circuit, USI: A-law or absent						
Values	COT: Continuity Indicator: co							
Comments	SIP-I		SU			ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>			<b>→</b>	СОТ		
	200 OK UPDATE	+						
			recondition	ons met				
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			<b>←</b>	ANM		
	ACK	<b>→</b>						
			Convers	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	<b>←</b>			<del>-</del>	RLC		

TP101040	SIP reference: RFC 3	3261 [4]			ISUP reference:			
				EN 383	3 001 [2], clause 6.1.3.5.2.2			
TSS reference		SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1/9	9					
criteria								
ISUP selection	NOT PICS 1/6 AND NOT PIC	CS 4/1						
criteria								
Test purpose	Ensure that the SUT on recei							
					ons in the SIP Supported header:			
	<ul> <li>the IAM shall be deferred</li> </ul>			litions have l	peen met;			
	<ul> <li>the u-law codec shall b</li> </ul>		<u>:d.</u>					
SIP parameter	Offer: m=audio 4711 RTF							
values	m=audio 4712 RTF	-						
	Answer: m=audio 0 RTP/A\	√P 0						
ISUP parameter								
values					1			
Comments	SIP-I		SU	Т	ISUP			
	INVITE(IAM)	<b>→</b>						
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	→		-	IAM			
	200 OK UPDATE	<b>←</b>						
			econditi	ons met				
	180 Ringing(ACM)	+		•	7.0			
	200 OK INVITE(ANM)	<b>←</b>		€	- ANM			
	ACK	→						
			Convers					
	BYE(REL)	→		-				
	200 OK BYE(RLC)	<b>←</b>		€	- RLC			

TP101041	SIP reference: RFC	3261 [4]			_	SUP reference:		
						01 [2], clause 6.1.3.5.2.2		
TSS reference		SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/						
SIP selection	PICS 4/4 AND PICS 4/5 ANI	D PICS 1	/9					
criteria								
ISUP selection	NOT PICS 1/6 AND NOT PI	CS 4/1						
criteria								
Test purpose	Ensure that the SUT on rece							
	a-law codec 100rel extension							
	<ul> <li>the IAM shall be deferred</li> </ul>			litions have	e bee	en met;		
	<ul> <li>the u-law codec shall</li> </ul>							
SIP parameter	Offer: m=audio 4711 RT	ΓΡ/AVP 0	)					
values	m=audio 4712 RT	TP/AVP 8						
	Answer: m=audio 0 RTP/A	AVP 0						
ISUP parameter								
values								
Comments	SIP-I		SU	JT		ISUP		
	INVITE(IAM)	→						
	183 Session Progress	<b>←</b>						
	PRACK	<b>→</b>						
	200 OK PRACK	<b>←</b>						
	UPDATE	<b>→</b>			<b>→</b>	IAM		
	200 OK UPDATE	+						
		F	reconditi	ions met				
	180 Ringing(ACM)	+			<b>←</b>	ACM		
	200 OK INVITE(ANM)	+			<b>←</b>	ANM		
	ACK	<b>→</b>						
			Conver	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			<del>(</del>	RLC		

TP101042	SIP reference: RFC 3261	S261 [4] ISUP reference:						
			EN:	383 001 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/ Sending of th	e Initial A	ddress Mes	sage (IAM)/				
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5	AND PI	S 1/9 AND	PICS 4/19				
criteria								
ISUP selection	NOT PICS 1/6							
criteria								
Test purpose	Ensure that the SUT on receipt of	an INVITE	message	with a SDP offer with more than				
	one media streams and based o	n operato	or policy th	en:				
	<ul> <li>the call is refused with a 415</li> </ul>	Unsupp	orted med	ia type response.				
SIP parameter	Offer: m=audio 4711 RTP/AVP 8							
values	m= audio 4712 RTP/AVP 8							
ISUP parameter								
values								
Comments	SIP-I		SUT	ISUP				
	INVITE(IAM)	NVITE(IAM) →						
	415 Unsupported media type	+	•					
	ACK	<b>→</b>	•					

TP101043	SIP reference: RFC 3261 [4] ISUP reference:								
		EN 383 001 [2], clause 6.1.3.5.2.2							
TSS reference	SIP-ISUP/Basic call/ Sending of the	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND PIC	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND PICS 4/19							
criteria									
ISUP selection	NOT PICS 1/6								
criteria									
Test purpose		Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than							
				ns extensions in the SIP Supported					
	header: and based on operator p								
	<ul> <li>the call is refused with a 41</li> </ul>	5 Unsup	orted medi	ia type response.					
SIP parameter	Offer: m=audio 4711 RTP/AVP 8								
values	m= audio 4712 RTP/AVP 8								
ISUP parameter									
values									
Comments	SIP-I		SUT	ISUP					
	INVITE(IAM) →								
	415 Unsupported media type ←								
	ACK	<b>→</b>							

TP101044	SIP reference: RFC 3261	[4]	EN 3	ISUP reference: 383 001 [2], clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sending of the	e Initial					
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND PICS						
ISUP selection criteria	NOT PICS 1/6						
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Require header and based on operator policy then:  • the call is refused with a 415 Unsupported media type response.						
SIP parameter	Offer: m=audio 4711 RTP/AVP 8			•			
values	m= audio 4712 RTP/AVP 8						
ISUP parameter values							
Comments	SIP-I		SUT	ISUP			
	INVITE(IAM)	<b>→</b>					
	415 Unsupported media type	+					
	ACK	<b>→</b>					

TP101045	SIP reference: RFC 3261 [4] ISUP reference:							
		EN 383 001 [2], clause 6.1.3.5.2.2						
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	NOT PICS 4/4 AND NOT PIC	CS 4/5 AND PIC	S 1/9 AND NO	Γ PICS 4/19				
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT on recei			a SDP offer with more than				
	one media streams and bas							
	<ul> <li>if the SDP offer contains</li> </ul>	one or more au	dio type media	streams and one or more				
			udio streams sl	nall be considered; the other				
	streams shall be rejected							
	<ul> <li>if the SDP offer contains</li> </ul>			ims, the IWU shall only				
	consider one, and reject		ns.					
SIP parameter	Offer: m=audio 4711 RTI							
values	m= audio 4712 RT							
	m= video 4713 RT	P/AVP 31						
	1							
	Answer: m=audio 4711 RTI							
	m=audio 0 RTP/A\							
10115	m=video 0 RTP/A\	VP 31						
ISUP parameter								
values	LOID I			LOUID				
Comments	SIP-I		UT	ISUP				
	INVITE(IAM)	<b>→</b>	<b>→</b>	IAM				
	180 Ringing(ACM)	<del>-</del>	+	ACM				
	200 OK INVITE(ANM)	+	+	ANM				
	ACK	<b>→</b>						
			rsation					
	BYE(REL)	<b>→</b>	<b>→</b>	REL				
	200 OK BYE(RLC)	<b>←</b>	<b>←</b>	RLC				

TP101046	SIP reference: RFC 3	3261 [4]			ISUP reference:			
					001 [2], clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19							
criteria								
ISUP selection	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1							
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Supported							
				onditions ex	tensions in the SIP Supported			
	header: and based on opera			B100 : 1	and the first state			
	the IAM shall be sent out     of Connection Indicators				with the coding of the Nature			
					streams and one or more			
					nall be considered; the other			
	streams shall be rejected		.,		,			
	<ul> <li>if the SDP offer contains</li> </ul>		audio type	media strea	ms, the IWU shall only			
	consider one, and reject				, , , , , , , , , , , , , , , , , , , ,			
SIP parameter	Offer: m=audio 4711 RTF							
values	m= audio 4712 RT	P/AVP 8	3					
	m= video 4713 RT	P/AVP 3	31					
	Answer: m=audio 4711 RTF	,						
	m=audio 0 RTP/A\							
ISUP parameter	m=video 0 RTP/A\ IAM: Continuity Indicator: <b>CO</b>		ovnoctod	LICI: A low	or abcont			
values	COT: Continuity Indicator: <b>co</b>			USI. A-law	DI ADSEIR			
Comments	SIP-I	I	SUT		ISUP			
	INVITE(IAM)	<b>→</b>	001	<b>→</b>	IAM			
	183 Session Progress	+			7,441			
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>		<b>→</b>	СОТ			
	200 OK UPDATE	+						
		Prec	onditions n	net				
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
			Conversa	tion				
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP101047	SIP reference: RFC	3261 [4]			ISUP reference:				
					3 001 [2], clause 6.1.3.5.2.2				
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/								
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19								
criteria									
ISUP selection	PICS 1/4 AND NOT PICS 1/6	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1							
criteria									
Test purpose		Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than							
		<b>one media streams</b> 100rel extensions and preconditions extensions in the SIP Require header <b>and based on operator policy then:</b>							
				a DICC aid	de with the coding of the Nature				
	of Connection Indicators					,			
					a streams and one or more				
					shall be considered; the other				
	streams shall be rejected		ny tino ada	io otroarrio	chan be concidered, and caner				
			audio type	media str	eams, the IWU shall only				
	consider one, and reject				, , , , , , , , , , , , , , , , , , ,				
SIP parameter	Offer: m=audio 4711 RTI								
values	m= audio 4712 RT								
	m= video 4713 RT	P/AVP (	31						
	Answer: m=audio 4711 RTI		3						
	m=audio 0 RTP/A								
ISUP parameter	m=video 0 RTP/A\ IAM: Continuity Indicator: CO		ovpostod	LICI: A los	v or about				
values	COT: Continuity Indicator: <b>co</b>			USI. A-Iav	v or absent				
Comments	SIP-I	l	, SUT	.	ISUP				
Commonto	INVITE(IAM)	<b>→</b>	001	-3					
	183 Session Progress	<del>-</del>							
	PRACK	<b>→</b>							
	200 OK PRACK	+							
	UPDATE	<b>→</b>		-	COT				
	200 OK UPDATE	+							
		Pred	conditions	met					
	180 Ringing(ACM)	+		€	- ACM				
	200 OK INVITE(ANM)	+		•	- ANM				
	ACK	<b>→</b>							
			Conversa	ation					
	BYE(REL)	<b>→</b>		-					
	200 OK BYE(RLC)	<b>←</b>		€	RLC				

TP101048	SIP reference: RFC 3	261 [4]			ISUP referenc	~ -		
					001 [2], clause	6.1.3.5.2.2		
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19							
criteria								
ISUP selection criteria	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1							
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than							
rest purpose	<b>one media streams</b> 100rel extensions and preconditions extensions in the SIP Support header: <b>and based on operator policy then:</b>							
	<ul> <li>the IAM shall be sent out immediately on the ISUP side with the Continuity of indicator "continuity check required on this circuit" or "continuity check performed on previous circuit";</li> </ul>							
	<ul> <li>if the SDP offer contains non-audio type media streams shall be rejected</li> </ul>	eam, on ;	ly the au	dio streams	shall be conside	red; the other		
	<ul> <li>if the SDP offer contains consider one, and reject to</li> </ul>				eams, the IWU sl	nall only		
SIP parameter	Offer: m=audio 4711 RTF		- ourourne	•				
values	m= audio 4712 RTI m= video 4713 RTI	P/AVP 8						
	Answer: m=audio 4711 RTF m=audio 0 RTP/AV m=video 0 RTP/AV	'P 8						
ISUP parameter	IAM: Continuity Indicator: con		check re	quired on t	his circuit or co	ntinuity check		
values	performed on previou							
	COT: Continuity Indicator: co							
Comments	SIP-I		SU	Т	ISUP			
	INVITE(IAM)	<b>→</b>		-	IAM			
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>		-	COT			
	200 OK UPDATE	+						
		Prec	onditions	met				
	180 Ringing(ACM)	+		€				
	200 OK INVITE(ANM)	+		€	- ANM			
	ACK	<b>→</b>						
		•	Convers	ation				
	BYE(REL)	<b>→</b>		-	REL			
	200 OK BYE(RLC)	+		+	- RLC			

TP101049	SIP reference: RFC 3261 [4]		ISUP reference:					
		EN 3	383 001 [2], clause 6.1.3.5.2.2					
TSS reference	SIP-ISUP/Basic call/ Sending of the Initial Address Message (IAM)/							
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND NOT PICS 4/19							
criteria								
ISUP selection	PICS 1/5 AND NOT PICS 1/6 AND PI	CS 4/1						
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Require header and based on operator policy then:  • the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "continuity check required on this circuit" or "continuity check performed on previous circuit";  • if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected;  • if the SDP offer contains several audio type media streams, the IWU shall only							
	consider one, and reject the other		oneding, the two shall only					
SIP parameter	Offer: m=audio 4711 RTP/AVP 8	ottoarno.						
values	m= audio 4712 RTP/AVP 8 m= video 4713 RTP/AVP 3  Answer: m=audio 4711 RTP/AVP 8 m=audio 0 RTP/AVP 8							
	m=video 0 RTP/AVP 31							
ISUP parameter	IAM: Continuity Indicator: continuity of							
values	performed on previous circulary COT: Continuity Indicator: continuity							
Comments	SIP-I	SUT	ISUP					
	INVITE(IAM) →		→ IAM					
	183 Session Progress ←							
	PRACK ->							
	200 OK PRACK ←							
	UPDATE →		→ COT					
	200 OK UPDATE ←							
	Preco	onditions met						
	180 Ringing(ACM) ←		<b>←</b> ACM					
	200 OK INVITE(ANM) ←		<b>←</b> ANM					
	ACK →							
		Conversation						
	BYE(REL) →		→ REL					
	200 OK BYE(RLC) ←		<b>←</b> RLC					

TP101050	SIP reference: RFC	3261 [4]			ISUP reference:	
					001 [2], clause 6.1.3.5.2.2	
TSS reference	SIP-ISUP/Basic call/ Sendir	ng of the Ini	tial Address M	1essage	· (IAM)/	
SIP selection	PICS 4/4 AND PICS 4/5 AN	ID PICS 1/9	AND NOT P	ICS 4/19	9	
criteria ISUP selection	NOT PICS 1/6 AND NOT P	100.4/4				
criteria	INOT PICS 1/6 AND NOT P	103 4/1				
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams 100rel extensions and preconditions extensions in the SIP Supported header: and based on operator policy then:  the IAM shall be deferred until all preconditions have been met;  if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected;  if the SDP offer contains several audio type media streams, the IWU shall only					
	consider one, and reject			ila Stica	inis, the ivve shall only	
SIP parameter values	Offer: m=audio 4711 RTP/AVP 8     m= audio 4712 RTP/AVP 8     m= video 4713 RTP/AVP 31  Answer: m=audio 4711 RTP/AVP 8     m=audio 0 RTP/AVP 8     m=video 0 RTP/AVP 31					
ISUP parameter values						
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	<b>→</b>				
	183 Session Progress	+				
	PRACK	<b>→</b>				
	200 OK PRACK	+				
	UPDATE	<b>→</b>		<b>→</b>	IAM	
	200 OK UPDATE	+				
		Preco	nditions met			
	180 Ringing(ACM)	+		<b>←</b>	ACM	
	200 OK INVITE(ANM)	+		+	ANM	
	ACK	→				
			Conversation	T		
	BYE(REL)	<b>→</b>		<b>→</b>	REL	
	200 OK BYE(RLC)	←		<b>←</b>	RLC	

TP101051	SIP reference: RF0	3261 [4]			ISUP reference:			
					01 [2], clause 6.1.3.5.2.2			
TSS reference	SIP-ISUP/Basic call/ Sendi	ng of the In	itial Address	Message	(IAM)/			
SIP selection	PICS 4/4 AND PICS 4/5 AN	ND PICS 1/	9 AND NOT F	PICS 4/19	9			
criteria	NOT BIOG 4/0 AND NOT B	100.4/4						
ISUP selection criteria	NOT PICS 1/6 AND NOT P	105 4/1						
Test purpose	Ensure that the SLIT on rec	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than						
	<ul> <li>one media streams 100rel extensions and preconditions extensions in the SIP Require header and based on operator policy then:</li> <li>the IAM shall be deferred until all preconditions have been met;</li> <li>if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected;</li> <li>if the SDP offer contains several audio type media streams, the IWU shall only</li> </ul>							
OID	consider one, and reject		streams.					
SIP parameter values	m= audio 4712 F m= video 4713 F Answer: m=audio 4711 R	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8 m= video 4713 RTP/AVP 31  Answer: m=audio 4711 RTP/AVP 8 m=audio 0 RTP/AVP 8						
ISUP parameter	III=Video 0 KTP//	AVF3I						
values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>						
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	→		<b>→</b>	IAM			
	200 OK UPDATE	+						
		Preco	onditions met					
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		<b>←</b>	ANM			
	ACK	→						
			Conversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		<b>+</b>	RLC			

## 5.2.1.2 Sending of the Subsequent Address Message (SAM)

TP102001	SIP reference: RFC	3261 [4]		ISUP reference:				
				12.5 [1], clause 6.2 a)				
TSS reference		SIP-ISUP/Basic call/ Sending of the Subsequent Address Message (SAM)/						
SIP selection	PICS 3/4							
criteria								
ISUP selection	PICS 3/8							
criteria								
Test purpose	Ensure that the SUT receives an INVITE with the same Call-ID and From tag as a previous INVITE which was associated with a BICC/ISUP call/bearer control instance currently existing on the BICC/ISUP side whereby the number of digits in the Request-URI is <b>greater</b> than the number of digits already accumulated for the call, sends a SAM and pass it to outgoing BICC/ISUP procedures.  The SAM shall contain in its Subsequent Number parameter only the additional digits received in this Request-URI compared with the digits already accumulated for the call.							
SIP parameter values								
ISUP parameter values	SAM; subsequent number	(PIXIT)						
Comments	SIP-I	S	UT	ISUP				
	INVITE	<b>→</b>	<b>→</b>	IAM				
	INVITE	<b>→</b>	<b>→</b>	SAM				
	INVITE	<b>→</b>	<b>→</b>	SAM				
	180 Ringing	+	+	ACM				
	200 OK INVITE	<del>-</del>	<b>←</b>	ANM				
	ACK	<b>→</b>						
		Conve	rsation					
	BYE(REL)	<b>→</b>	<b>→</b>	REL				
	200 OK BYE(RLC)	+	+	RLC				

TP102002	SIP reference: RFC 3	261 [4]		-	SUP reference:			
	Q.1912.5 [1], clause 6.2 b)							
TSS reference	SIP-ISUP/Basic call/ Sending	of the Subsec	uent Addres	ss Me	ssage (SAM)/			
SIP selection	PICS 3/4							
criteria								
ISUP selection criteria	PICS 3/8							
Test purpose	Ensure that the SUT receives an INVITE with the same Call-ID and From tag as a previous INVITE which was associated with a BICC/ISUP call/bearer control instance currently existing on the BICC/ISUP side whereby the number of digits in the Request-URI is <b>fewer</b> than the number of digits already accumulated for the call:  • then the SUT shall immediately send a <b>484 Address Incomplete</b> response for this INVITE;  • in this case no SAM is sent to BICC/ISUP procedures.							
SIP parameter values								
ISUP parameter								
values								
Comments	SIP-I	,	SUT		ISUP			
	INVITE(IAM)	<b>→</b>	·	<b>→</b>	IAM			
	INVITE(IAM)	<b>→</b>		•				
	484 Address incomplete	+		<b>→</b>	REL			
	ACK	<b>→</b>		+	RLC			

## 5.2.1.3 Sending of COT

TP103001	SIP reference: RF	C 3261 [4	]		ISUP reference: 912.5 [1], clause 6.3			
TSS reference	SIP-ISUP/Basic call/COT	SIP-ISUP/Basic call/COT						
SIP selection criteria	PICS 4/4 AND PICS 4/5							
ISUP selection criteria	PICS 1/4 AND PICS 4/1	PICS 1/4 AND PICS 4/1						
Test purpose	Ensure that the when the SUT determines that all the preconditions on the incoming SIP side have been met and any continuity procedures on the outgoing BICC side have been successfully completed:  the SUT shall send the COT message where the Continuity Indicator in the COT message shall be set to "Continuity".							
SIP parameter values								
ISUP parameter values	COT continuity indicator: C	ontinuity	7					
Comments	SIP-I		SU	Т	ISUP			
	INVITE(IAM)	<b>→</b>		→	IAM			
	183 Session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>		<b>→</b>	СОТ			
	200 OK UPDATE	+						
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
			Convers	ation				
	BYE(REL)	<b>→</b>		→	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP103002	SIP reference: RFC 3261 [4]				ISUP reference:			
				Q.19	912.5 [1], clause 6.3			
TSS reference	SIP-ISUP/Basic call/ COT	SIP-ISUP/Basic call/ COT						
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISUP selection	PICS 1/5 AND PICS 4/1							
criteria								
Test purpose	Ensure that the when the SUT determines that all the preconditions on the incoming SIP side have been met and any continuity procedures on the outgoing ISUP side have been successfully completed:  • the I-IWU shall send the COT message where the Continuity Indicator in the COT message shall be set to "Continuity check successful".							
SIP parameter								
values								
ISUP parameter	COT continuity indicator: Co	ontinuity c	heck suc	cessful;				
values								
Comments	SIP-I		SU		ISUP			
	INVITE(IAM)	→		→	IAM			
	183 Session Progress	+						
	PRACK	→						
	200 OK PRACK	<b>←</b>						
	UPDATE	→		→	COT			
	200 OK UPDATE	<b>←</b>						
		Р	reconditi	ons met				
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	ACK	<b>→</b>						
			Convers	sation				
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

## 5.2.1.4 Receipt of the Address Complete Message (ACM)

TP104001	SIP reference: RFC 326	1 [4]		-	SUP reference:	
					2.5 [1], clause 6.5 2)	
TSS reference	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Ensure that the SUT on receipt of an ACM message where the Called party status indicator is set to "no indication":  183 Session Progress response is sent from the I-IWU;  the received ACM is encapsulated in the 183 Session Progress.					
SIP parameter values						
ISUP parameter values	ACM Called party status: no indication;					
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM	
	183 Session Progress (ACM)	+		+	ACM(no indication)	
	200 OK INVITE(ANM)	+		+	ANM	
	ACK	<b>→</b>				
		Co	nversation			
	BYE(REL)	<b>→</b>		<b>→</b>	REL	
	200 OK BYE(RLC)	+		+	RLC	

TP104002	SIP reference: RFC 3261 [4]			-	SUP reference: 2.5 [1], clause 6.5 1)	
TSS reference	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/					
SIP selection criteria						
ISUP selection criteria						
Test purpose	<ul> <li>Ensure that the SUT on receipt of an ACM message where the Called party status indicator is set to "subscriber free" where the ISUP indicator parameter set to ISUP_ID, the ISDN access indicator set to ISDN_ACCES_ID and the OBCI in-band information set to OBCI_INBAND then:</li> <li>the 180 Ringing SIP response is sent. Ensure that the in-band information can be transmitted to the calling user;</li> <li>the received ACM is encapsulated in the 180 Ringing.</li> </ul>					
SIP parameter values		'				
ISUP parameter values	ACM FCI: ISUP_ID, ISDN_ACCESS_ID, OBCI: OBCI_INBAND;					
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM	
	180 Ringing(ACM)	+		+	ACM	
	200 OK INVITE(ANM)	+		+	ANM	
	ACK	<b>→</b>				
			Conversation			
	BYE(REL)	<b>→</b>		<b>→</b>	REL	
	200 OK BYE(RLC)	+		+	RLC	

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_ACCES_ID: ISDN
	OBCI_INBAND: yes

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

TP105001	SIP reference: RFC 3261 [4]				ISUP reference: 912.5 [1], clause 6.6
TSS reference	SIP-ISUP/Basic call/ Receipt of	the Cal	l progress me	ssage (0	CPG).
SIP selection criteria					
ISUP selection					
criteria					
Test purpose	Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the <b>event information parameter event indicator</b> is set to "Alerting":  the 180 Ringing SIP response is sent;  The received CPG is encapsulated in the 180 Ringing.				
SIP parameter					
values					
ISUP parameter	ACM: Called party status "no inc	dication	"		
values	CPG; event information param	neter ev	ent indicator	r: Alertin	ng
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	183 Session Progress (ACM)	+		+	ACM(no indication)
	180 Ringing(CPG)	+		+	CPG(ALERTING)
	200 OK INVITE(ANM)	+		+	ANM
	ACK	<b>→</b>			
			Conversation	1	
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP105002	SIP reference: RFC 326	1 [4]			ISUP reference: 912.5 [1], clause 6.6		
TSS reference	SIP-ISUP/Basic call/ Receipt of t	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).					
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Ensure that the SUT, having received the ACM message, on receipt of a CPG message where the <b>event information parameter event indicator</b> is set to "Progress":  183 Session Progress response is sent from the I-IWU; the received CPG is encapsulated in the 183 Session Progress.						
SIP parameter							
values							
ISUP parameter	ACM: Called party status "no ind	ication'	1				
values	CPG; event information param	eter ev	ent indicator: I	Progre	ess		
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	183 Session Progress (ACM)	+		+	ACM(no indication)		
	183 Session (CPG)	+		+	CPG(PROGRESS)		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	<b>→</b>					
			Conversation				
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP105003	SIP reference: RFC 3261 [4]				ISUP reference:
				Q.1	912.5 [1], clause 6.6
TSS reference	SIP-ISUP/Basic call/ Receipt of	the Cal	l progress mes	sage (	CPG).
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Ensure that the SUT, having rec				
	where the event information pa			<b>ator</b> is	set to "in-band information or
	an appropriate pattern is now				
	<ul> <li>183 Session Progress response</li> </ul>	onse is	sent from the	-IWU;	
	<ul> <li>the received CPG is encaps</li> </ul>	sulated	in the 183 -ses	sion P	rogress.
SIP parameter					
values					
ISUP parameter	ACM: Called party status "no inc				
values	CPG; event information param	neter ev	ent indicator	in-bar	nd-information or an appropriate
	pattern is now available				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	183 Session Progress (ACM)	+		+	ACM(no indication)
	183 Session (CPG)	+		+	CPG (Inbad Info available)
	200 OK INVITE(ANM)	+		+	ANM
	ACK	<b>→</b>			
			Conversation	•	
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

## 5.2.1.6 Receipt of the Answer message (ANM)

TP106001	SIP reference: RFC 32	61 [4]		ISUP reference:		
				Q.1912.5 [1], clause 6.7		
TSS reference	SIP-ISUP/Basic call/ Receipt of	the Answer	message (ANI	M).		
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose		Ensure that the SUT, having received the ACM message, on receipt of an ANM message:				
	<ul> <li>sends a 200 OK INVITE;</li> </ul>					
	the received ANM is encar					
		cted in both	directions whe	n both of the following conditions		
	are satisfied:					
				-T Recommendation Q.1902.4		
	[Error! Reference source					
				n RFC 3312 [5]) that sufficient		
		atisfied on th	e SIP side for	session establishment to proceed		
	(if applicable).	4h a "Dan a		un in the formulation discosticus!		
	In addition, if BICC is performing			s " <b>notification not required</b> ", the		
	bearer path shall be connected					
		and the I-IWU determines (through the procedures defined in RFC 3312 [5]) that sufficient preconditions have been met for the session to proceed.				
SIP parameter	200 OK INVITE with encapsula		rto proceed.			
values	200 OK IIVITE With Chapsula	200 OK INVITE With encapsulated Alvivi				
ISUP parameter	ANM					
values	, t					
Comments	SIP-I		SUT	ISUP		
	INVITE(IAM)	<b>→</b>	-	<b>→</b> IAM		
	180 Ringing(ACM)	+	•	E ACM		
	200 OK INVITE(ANM)	+	•	E ANM		
	ACK	<b>→</b>				
		Conv	ersation			
	BYE(REL)	<b>→</b>	-	<b>→</b> REL		
	200 OK BYE(RLC)	+	•	F RLC		

#### 5.2.1.7 Receipt of the Connect message (CON)

TP107001	SIP reference: RFC 320	61 [4]		ISUP reference:		
				2.5 [1], clause 6.4, 6.7		
TSS reference	SIP-ISUP/Basic call/ Receipt of	the CONNEC	T message (CC	DN).		
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	SDP offer was received in the initial INVITE. Ensure that the SUT, on receipt of an CON					
	message:					
	sends a 200 OK INVITE;		000 01/ 15 11/17			
	the received CON is encape  The shade and the shade are shaded as the shade are shaded as the shade are shaded as the shaded are shaded as th					
	The bearer path shall be connectate satisfied:	ctea in both a	rections when t	ooth of the following conditions		
				Recommendation Q.1902.4		
	<ul> <li>[Error! Reference source not found.]) is successfully completed; and</li> <li>the I-IWU determines (using the procedures defined in RFC 3312 [5]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable).</li> <li>In addition, if BICC is performing the "Per-call bearer set-up in the forward direction"</li> <li>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 [5]) that sufficient</li> </ul>					
	preconditions have been met for the session to proceed.					
SIP parameter						
values						
ISUP parameter						
values		•	•			
Comments	SIP-I		UT	ISUP		
		<b>→</b>	<b>→</b>	IAM		
	( /	<del>(</del>	+	CON		
	ACK	<b>→</b>	···			
	DVE(DEL)		rsation	DEL		
	- : - (: : /	<del>)</del>	<b>→</b>	REL		
	200 OK BYE(RLC)	<b>←</b>	<b>←</b>	RLC		

#### 5.2.1.8 Receipt of the REL message

TP108001	SIP reference: RFC 3	SIP reference: RFC 3261 [4] ISUP reference:				
				Q.191	2.5 [1], clause 6.11.2	
TSS reference	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/					
SIP selection						
criteria						
ISUP selection criteria						
Test purpose  SIP parameter values	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:  • 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 with the encapsulated REL message.  SIP Statue-Code: SIP_FAILURE_VA (PIXIT)					
ISUP parameter values	REL; cause value: CV_ISUP (PIXIT)					
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM	
	SIP_FAILURE_VA(REL)	+		+	REL	
	ACK	<b>→</b>		<b>→</b>	RLC	

Table 1

	Values for test purpose TP108001				
	←SIP Message SIP_FAILURE_VA	← REL Cause Indicators parameter CV_ISUP			
VA_1	404 Not Found	Cause Value No. 1 ("unallocated (unassigned) number")			
VA_2	500 Server Internal Error	Cause Value No. 2 ("no route to network")			
VA_3	500 Server Internal Error	Cause Value No. 3 ("no route to destination")			
VA_4	500 Server Internal Error	Cause Value No. 4 ("Send special information tone")			
VA_5	404 Not Found	Cause Value No. 5 ("Misdialled trunk prefix")			
VA_6	500 Server Internal Error	Cause Value No. 8 ("Pre-emption")			
VA_7	500 Server Internal Error	Cause Value No. 9 ("Pre-emption-circuit reserved for reuse")			
VA_8	486 Busy Here	Cause Value No. 17 ("user busy")			
VA_9	480 Temporarily unavailable	Cause Value No. 18 ("no user responding")			
VA_10	480 Temporarily unavailable	Cause Value No. 19 ("no answer from the user")			
VA_11	480 Temporarily unavailable	Cause Value No. 20 ("subscriber absent")			
VA_12	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")			
VA_13	410 Gone	Cause Value No. 22 ("number changed")			
VA_14	480 Temporarily unavailable	Cause Value No. 25 ("Exchange routing error")			
VA_15	502 Bad Gateway	Cause Value No. 27 ("destination out of order")			
VA_16	484 Address Incomplete	Cause Value No. 28 ("invalid number format (address			
		incomplete")			
VA_17	500 Server Internal Error	Cause Value No. 29 ("facility rejected")			
VA_18	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified")			
_	,	(Class default)			
VA_19	486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible)	Cause Value in the Class 010 (No circuit/channel available, Cause Value No. 34)			
\/A 00	else 480 Temporarily unavailable 500 Server Internal Error	Course Value in the Class 040 (recourse unavailable Course			
VA_20	500 Server Internal Error	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 ("requested facility not subscribed")			
VA_21	500 Server Internal Error (SIP-I only)	Cause Value No. 55 ("incoming calls barred within CUG")			
VA_23	500 Server Internal Error	Cause Value No. 57 ("bearer capability not authorized")			
VA_24	500 Server Internal Error	Cause Value No. 58 ("bearer capability not presently")			
VA_25	500 Server Internal Error	Cause Value No. 63 ("service option not available, unspecified") (Class default)			
VA_26	500 Server Internal Error	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 ("user not member of CUG")			
VA_28	500 Server Internal Error	Cause Value No. 88 ("incompatible destination")			
VA 29	500 Server Internal Error	Cause Value No. 90 ("Non-existent CUG")			
VA_30	404 Not Found	Cause Value No. 91 ("invalid transit network selection")			
VA_31	500 Server Internal Error	Cause Value No. 95 ("invalid message") (Class default)			
VA_32	500 Server Internal Error	Cause Value No. 97 ("Message type non-existent or not implemented")			
VA_33	500 Server Internal Error	Cause Value No. 99 ("information element/parameter non-existent or not implemented")			
VA_34	480 Temporarily unavailable	Cause Value No. 102 ("recovery on timer expiry")			
VA_35	500 Server Internal Error	Cause Value No. 103 ("Parameter non-existent or not implemented, pass on")			
VA_36	500 Server Internal Error	Cause Value No. 110 ("Message with unrecognized Parameter, discarded")			
VA_37	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)			
VA_38	480 Temporarily unavailable	Cause Value No. 127 ("interworking unspecified") (Class default)			

TP108002	SIP reference: RFC 32	61 [4]		Q		SUP reference: 2.5 [1], clause 6.11.2	
TSS reference	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/						
SIP selection criteria							
ISUP selection criteria							
Test purpose	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:  • 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 with the encapsulated REL message.						
SIP parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT)						
ISUP parameter values	REL; cause value: CV_ISUP (I	PIXIT)					
Comments	SIP-I		SUT			ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	183 Session Progress(ACM)	+			<b>←</b>	ACM(no indication)	
	SIP_FAILURE_VA(REL)	+			<del>-</del>	REL	
	ACK	<b>^</b>			<b>→</b>	RLC	

Table 2

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

TP108003	SIP reference: RFC 3	261 [4]			ı	SUP reference:	
				(	Q.191	2. [1],5 clause 6.11.2	
TSS reference	SIP-ISUP /Basic call/ Receipt	of the R	elease n	nessage	(REL)	/	
SIP selection criteria							
ISUP selection criteria							
Test purpose	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:  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 with the encapsulated REL message.						
SIP parameter values	SIP Statue-Code: SIP_FAILUR	E_VA (P	IXIT)				
ISUP parameter values	REL; cause value: CV_ISUP	(PIXIT)					
Comments	SIP-I		SU	Т		ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	180 Ringing(ACM)	+			+	ACM	
	SIP_FAILURE_VA(REL)	+			+	REL	
	ACK	<b>→</b>			<b>→</b>	RLC	

TP108004	SIP reference: RFC 32	61 [4]			ı	SUP reference:	
				(	Q.191	2.5 [1], clause 6.11.2	
TSS reference	SIP-ISUP /Basic call/ Receipt o	f the R	elease r	nessage	(REL)	/	
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	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 ISUP REL:  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 with the encapsulated REL message.						
SIP parameter values	SIP Statue-Code: SIP_FAILURE	_VA (PI	XIT)				
ISUP parameter	REL; cause value: CV_ISUP (I	PIXIT)					
values							
Comments	SIP-I		SI	JT		ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	183 Session Progress(ACM)	<b>←</b>			+	ACM(no indication)	
	180 Ringing(CPG)	+			+	CPG(ALERTING)	
	SIP_FAILURE_VA(REL)	+			+	REL	
	ACK	<b>→</b>			<b>→</b>	RLC	

Table 3

	Values for test purposes TP108003 and TP108004					
←SIP Message SIP_FAILURE_VA		← REL Cause Indicators parameter CV_ISUP,				
VA_1	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")				
VA_2	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified") (Class default)				
VA_4	500 Server Internal Error	Cause Value No. 38 ("Network out of order")				
VA_4	500 Server Internal Error	Cause Value No. 41 ("Temporary failure ")				
VA_5	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)				

TP108005	SIP reference: RFC	3261 [4]		•	SUP reference:		
				Q.191	2.5 [1], clause 6.11.2		
TSS reference	SIP-ISUP /Basic call/ Receip	ot of the Re	elease messa	ge (REL)	/		
SIP selection criteria							
ISUP selection criteria							
Test purpose	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 a ANM", a 200 OK message is sent, on receipt of an ISUP REL, where the cause value defined as <b>CV_ISUP</b> :  • 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 a BYE message with the encapsulated REL message.						
SIP parameter values							
ISUP parameter							
values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	<b>→</b>					
			Conversation				
	BYE(REL)	+		+	REL		
	200 OK BYE(RLC)	<b>→</b>		<b>→</b>	RLC		

TP108006	SIP reference: RFC	3261 [4	]		ISUP reference: 2.5 [1], clause 6.11.2			
TSS reference	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/							
SIP selection criteria								
ISUP selection criteria								
Test purpose	message, having received a ISUP REL, where the cause the SUT immediately reconsults available	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an 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 <b>CV_ISUP</b> :  • 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 a BYE message with the encapsulated REL message.						
SIP parameter values								
ISUP parameter values	REL; cause value: CV_ISUF	P (PIXIT	()					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	200 OK INVITE(CON)	+		+	CON			
	ACK	<b>→</b>						
			Conversation	1				
	BYE(REL)	+		+	REL			
	200 OK BYE(RLC)	<b>→</b>		<b>→</b>	RLC			

Table 4

	Values for test purpose TP108005 and TP 108006					
←SIP Message SIP_FAILURE_VA		← REL Cause Indicators parameter CV_ISUP				
VA_1	BYE	Cause Value No. 16				
VA_2	BYE	Cause Value No. 31 ("normal unspecified") (Class default)				
VA_3	BYE	Cause Value No. 38 ("Network out of order")				
VA_4	BYE	Cause Value No. 41 ("Temporary failure ")				
VA_5	BYE	Cause Value No. 111 ("protocol error, unspecified") (Class default)				

TP108007	SIP reference: RF	C 3261 [4]			Į	SUP reference:
					Q.191	2.5 [1], clause 6.11.2
TSS reference	SIP-ISUP /Basic call/ Rece	ipt of the F	Release n	nessage	(REL)	/
SIP selection	PICS 4/21					
criteria						
ISUP selection						
criteria						
Test purpose	message, on receipt of an	ISUP REL requests th	with cau	se value	23 the	nessage, sending out an IAM SUT shall: destination according the
SIP parameter values						
ISUP parameter	REL; cause value: 23					
values						
Comments	SIP-I		SU	T		ISUP
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM(Destination 1)
					+	REL(new Destination)
					<b>→</b>	RLC
					<b>→</b>	IAM(Destination 2)
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	<b>←</b>			<b>←</b>	ANM
	ACK	<b>→</b>				
			Convers	sation		
	BYE(REL)	+			+	REL
	200 OK BYE(RLC)	<b>→</b>			<b>→</b>	RLC

#### 5.2.1.9 Autonomous release at I-IWU

TP109001	SIP reference: RFC 3261 [4]				ISUP reference: I2.5 [1], clause 6.11.3		
TSS reference	SIP-ISUP/Basic call/ Autonomous rele	ase a		Q.13	12.5 [1], Clause 0.11.5		
SIP selection criteria							
ISUP selection criteria	PICS 4/6						
Test purpose	(because the call is not routable), the	Ensure that when a an automatic repeat attempt initiated by the SUT is not successful (because the call is not routable), the SUT shall:  • send a 480 Temporarily Unavailable response to the SIP side. No actions on the ISUP (BICC) side are required.					
SIP parameter values							
ISUP parameter values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		<b>→</b>	IAM		
	480 Temporarily unavailable (REL)	+		+	RSC		
	ACK	<b>→</b>		<b>→</b>	RLC		

TP109002	SIP reference: RFC 3261	[4]			ISUP reference:		
				Q.191	12.5 [1], clause 6.11.3		
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU						
SIP selection criteria							
ISUP selection							
criteria							
Test purpose	information and determines that the	Ensure that when the SUT receives unrecognized backward ISUP or BICC signalling nformation and determines that the call needs to be released based on the coding, the SUT:  shall send a 500 Server Internal Error response on the SIP side.					
SIP parameter values							
ISUP parameter values	Unknown message: Message com	patibility "F	Release c	all"			
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
				+	???		
	500 Server internal error(REL)	+		<b>→</b>	REL		
	ACK	<b>→</b>		+	RLC		

TP109003	SIP reference: RFC 3261	[4]		ISUP reference: Q.1912.5 [1], clause 6.11.3				
TSS reference	SIP-ISUP/Basic call/ Autonomous	SIP-ISUP/Basic call/ Autonomous release at I-IWU						
SIP selection criteria								
ISUP selection criteria	PICS 3/4							
Test purpose	<ul><li>Ensure that the SUT on receipt of</li><li>sends an 484 Address Incom</li></ul>		3	ceived in an INVITE messages:				
SIP parameter values			•					
ISUP parameter values								
Comments	SIP-I		SUT	ISUP				
	484 Address incomplete	→ ← →						

TP109004	SIP reference: RFC 32	261 [4]		ISUP reference:		
			Q.19 <sup>4</sup>	12.5 [1], clause 6.11.3		
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU					
SIP selection	PICS 3/4					
criteria						
ISUP selection						
criteria						
Test purpose	Ensure that the SUT on receipt	t of subsequent	INVITE messa	ge:		
SIP parameter	is sending a 484 Address Incomplete message to consider any offer-answer exchange initiated by the INVITE. A new INVITE shall initiate a new offer-answer exchange.  As a general principle, the overlap procedures allow for session negotiation (and in particular the negotiation and confirmation of preconditions) to continue independently of the receipt of address information. On sending of a 484 Address Incomplete message for an INVITE transaction the I-IWU considers any offer-answer exchange initiated by the INVITE to be terminated. The new INVITE initiates a new offer-answer exchange. However, if resources have already been reserved and they can be reused within the new offer-answer exchange, the precondition signalling shall reflect the current status of the affected preconditions.					
SIP parameter values						
ISUP parameter						
values						
Comments	SIP-I	SI	JT	ISUP		
	INVITE(IAM)	<b>→</b>				
	INVITE(IAM)	<b>→</b>				
	484 Address incomplete	+				
	ACK	<b>→</b>				

TP109005	SIP reference: RFC 326	1 [4]		ISUP reference: Q.1912.5 [1], clause 6.11.3	
TSS reference	SIP-ISUP/Basic call/ Autonomou	ıs releas	e at I-IWU		
SIP selection criteria					
ISUP selection criteria					
Test purpose	<ul><li>Ensure that the SUT in congestion</li><li>sends an 480 Temporarily Ut</li></ul>		•	message:	
SIP parameter values			•		
ISUP parameter values					
Comments	SIP-I		SUT	ISUP	
	INVITE(IAM)	<b>→</b>			
	480 Temporarily unavailable	+			
	ACK	<b>→</b>			

TP109006	SIP reference: RFC 3261 [4] ISUP reference:								
	Q.1912.5 [1], clause 6.11.3								
TSS reference	SIP-ISUP/Basic call/ Autonomous release	at I-IWU							
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the call is released due to the I	BICC/ISUP con	mpatibility procedure for unknown						
	parameters:								
	<ul> <li>sends 500 Server Internal Error.</li> </ul>								
SIP parameter									
values									
ISUP parameter	Unknown parameter in ACM: Parameter co	mpatibility "Re	elease call"						
values									
Comments	SIP-I	SUT	ISUP						
	INVITE(IAM) →		→ IAM						
		•	← ACM(???)						
	500 Server internal error(REL) ←		→ REL						
	ACK →	•	<b>←</b> RLC						

TP109007	SIP reference: RFC 32	61 [4]		ISUP reference: Q.1912.5 [1], clause 6.11.3					
TSS reference	SIP-ISUP/Basic call/ Autonomous release at I-IWU								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the call is released	due to	expiry of T7 w	vithin the	BICC/ISUP procedures:				
	<ul> <li>sends 484 Address Incomp</li> </ul>	olete.			·				
SIP parameter									
values									
ISUP parameter									
values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
			T7 expiry	•					
	484 Address incomplete	+		<b>→</b>	REL				
	ACK	<b>→</b>		+	RLC				

TP109008	SIP reference: RFC 32	61 [4]		ISUP reference: Q.1912.5 [1], clause 6.11.3					
TSS reference	SIP-ISUP/Basic call/ Autonomo	us rele	ase at I-IWU						
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose		Ensure that the call is released due expiry of T9 within the BICC/ISUP procedures:  • sends 480 Temporarily Unavailable.							
SIP parameter values									
ISUP parameter values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
		T9 expiry							
	480 Temporarily unavailable	<b>←</b>		<b>→</b>	REL				
	ACK	<b>→</b>		+	RLC				

## 5.2.1.10 Receipt of the Release message BYE / CANCEL

TP110001	SIP reference: RF0	C 3261 [4]			ISUP reference: I2.5 [1], clause 6.11.1				
TSS reference	SIP-ISUP/Basic call/ Receipt of the BYE-CANCEL message								
SIP selection criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT on rec	eipt of SII	PBYE, the SU	JT shall se	end an ISUP REL with the				
	cause value # 16 to the ISL	JP side.							
SIP parameter									
values									
ISUP parameter values	REL: Cause value #16, Loc	ation "Ne	twork beyond a	an interwo	rking point"				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
	ACK	→							
			Conversation	1					
	BYE(REL)	→		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP110002	SIP reference: RFC			IS	SUP reference:				
	Q.1912.5 [1], clause 6.11.1								
TSS reference	SIP-ISUP/Basic call/ Receipt of the BYE-CANCEL message								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT on rece	ipt of SIP	CANCE	L, the I-IWU	J sha	all send an ISUP REL with the			
	cause value # 31 to the ISUF	o side.							
SIP parameter	CANCEL without encapsulat	ed ISUP	message	)					
values									
ISUP parameter	REL: Cause value #31, Loca	tion "Net	work bey	ond an inte	rwor	king point"			
values									
Comments	SIP-I		SU	IT		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	+		•	<del>(</del>	ACM			
	CANCEL	<b>→</b>			<b>→</b>	REL			
	200 OK CANCEL	200 OK CANCEL ← RLC							
	487 Request Terminated	+							
	ACK	<b>→</b>							

TP110003	SIP reference: RFC 3261 [4]				ISUP reference: I2.5 [1], clause 6.11.1				
TSS reference	SIP-ISUP/Basic call/ Receipt of the BYA-CANCEL message								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT on rece	•	BYE, the I-	-IWU shall s	end an ISUP REL with the				
	cause value # 31 to the ISUF								
SIP parameter	BYE without encapsulated IS	SUP mess	age						
values									
ISUP parameter values	REL: Cause value #31, Loca	ition "Netw	ork beyon	d an interwo	rking point"				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	BYE	<b>→</b>		<b>→</b>	REL				
	200 OK BYE	+		+	RLC				
	487 Request Terminated	+							
	ACK	<b>→</b>	•						

# 5.2.1.11 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 [4]	C	ISUP reference: Q.1912.5 [1], clause 6.11.4 and 5						
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	already been received on re	a BTE moodage if the GOT has already received an Mortion the 200 GR in VITE moodage								
SIP parameter values										
ISUP parameter values										
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM					
	180 Ringing(ACM)	<del>-</del>		+	ACM					
	200 OK INVITE(ANM)	<del>-</del>		+	ANM					
	ACK									
		С	onversation	•						
	BYE(REL)	<del>(</del>		+	RSC					
	200 OK BYE(RLC)	<b>→</b>		<b>→</b>	RLC					

TP111002	SIP reference: RFC 3	3261 [4]			I	SUP reference:			
	Q.1912.5 [1], clause 6.11.4 and 5								
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message								
	(GRS) or Circuit group blocking	ng messa	ige (CGE	B) with the	indica	tion hardware failure oriented			
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose						nessage relating to the call has			
	already been received on rec								
	<ul> <li>a BYE message if the SU which had it sent.</li> </ul>	JT has alr	ready red	eived an	ACK fo	or the 200 OK INVITE message			
SIP parameter									
values									
ISUP parameter									
values									
Comments	SIP-I		S	UT		ISUP			
	INVITE(IAM)	→			→	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	ACK →							
			Conve	rsation					
	BYE	+		_	+	GRS			
	200 OK BYE	<b>→</b>			<b>→</b>	GRA			

TP111003	SIP reference: RFC 3261 [4] ISUP reference:								
	Q.1912.5 [1], clause 6.11.4								
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message								
	(GRS) or Circuit group blocking	messa	age (CGB)	with the	indica	tion hardware failure oriented			
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	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 BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent.								
SIP parameter values									
ISUP parameter values	Circuit Group Supervision Mess	sage Ty	pe Indicate	or "hard	ware f	ailure oriented"			
Comments	SIP-I		SU	Γ		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
	ACK	<b>→</b>							
			Convers	ation					
	BYE	+			+	CGB(hardware failure)			
	200 OK BYE	<b>→</b>		•	<b>→</b>	CGBA			

TP111004	SIP reference: RFC	3261 [4]		I	SUP reference:			
	Q.1912.5 [1], clause 6.11.4 and 5							
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message							
	(GRS) or Circuit group blocki	ing message (Co	GB) with the	indica	tion hardware failure oriented			
SIP selection								
criteria								
ISUP selection criteria								
Test purpose					message relating to the call has			
	already been received on rec							
	200 OK INVITE if the SUT ha							
	<ul> <li>the SUT shall wait until in BYE.</li> </ul>	t receives the A0	CK for the 20	00 OK	INVITE before sending the			
SIP parameter								
values								
ISUP parameter								
values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	←		+	ACM			
	200 OK INVITE(ANM)	←		+	ANM			
				+	RSC			
	ACK	<b>→</b>		<b>→</b>	RLC			
	BYE(REL)	+						
	200 OK BYE(RLC)	<b>→</b>						

TP111005	SIP reference: RFC 32	261 [4]		0.1	_	SUP reference:		
TSS reference:	Q.1912.5 [1], clause 6.11.4 and 5 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:	(	<u> </u>	J = \					
ISUP selection criteria:								
Test purpose:	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 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE:  • the SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE.							
SIP parameter values:								
ISUP parameter values:								
Comments:	SIP-I		5	SUT		ISUP		
	INVITE(IAM)	→			<b>→</b>	IAM		
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
	€ GRS							
	ACK	<b>→</b>			<b>→</b>	GRA		
	BYE	+						
	200 OK BYE	<b>→</b>						

TP111006	SIP reference: RFC 32	261 [4]		ISUP reference:					
	Q.1912.5 [1], clause 6.11.4								
TSS reference	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message								
	(GRS) or Circuit group blocking	g message (CG	B) with the indic	ation hardware failure oriented					
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT, when at I								
	already been received on received								
	Message Type Indicator coded								
	200 OK INVITE if the SUT has								
		eceives the AC	K for the 200 Of	CINVITE before sending the					
CID noremeter	BYE.								
SIP parameter values									
	Circuit Croup Supervision Mos	aaga Tupa Indi	ootor "bordwore	failure oriented"					
ISUP parameter values	Circuit Group Supervision Mes	sage Type Indi	cator naroware	ranure oriented					
Comments	SIP-I	1 1	SUT	ISUP					
Comments	INVITE(IAM)	<b>→</b>	→	IAM					
	180 Ringing(ACM)	<del>-</del>	· ·	ACM					
	200 OK INVITE(ANM)	+	÷	ANM					
	200 01(114/112(/114///)	+		/ ((VIV)					
			+	CGB(hardware failure)					
	ACK	<b>→</b>	→	CGBA					
	BYE	+							
	200 OK BYE	<b>→</b>							

TP111007	SIP reference: RFC 3261	[4]	Q.	ISUP reference: Q.1912.5 [1], clause 6.11.4 and 5			
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	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a RSC message sends:  a 500 Server Internal Error on the SIP side.						
SIP parameter values							
ISUP parameter values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	500 Server Internal Error(REL)	+		<del>-</del>	RSC		
	ACK	<b>→</b>		<b>→</b>	RLC		

TP111008	SIP reference: RFC 326	61 [4]	Q.	_	SUP reference: [1], clause 6.11.4 and 5	
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	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends:  • a 500 Server Internal Error on the SIP side.					
SIP parameter values						
ISUP parameter values						
Comments	SIP-I INVITE(IAM)	<b>→</b>	SUT	<b>→</b>	ISUP IAM	
	180 Ringing(ACM) 500 Server Internal Error	<del>+</del>		<del>+</del>	ACM	
	GRS GRA					

TP111009	SIP reference: RFC 326	61 [4]		(		SUP reference: 2.5 [1], clause 6.11.4	
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	Ensure that the SUT, when at least one backward ISUP message relating to the call has already been received on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented", sends:  • a 500 Server Internal Error on the SIP side.						
SIP parameter							
values							
ISUP parameter							
values							
Comments	SIP-I		SI	JT		ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	180 Ringing(ACM)	4			+	ACM	
	500 Server Internal Error	4			+	CGB(hardware failure)	
	ACK	<b>→</b>			<b>→</b>	CGBA	

TP111010	SIP reference: RFC 3261 [4]		ISUP reference:			
	Q.1912.5 [1], clause 6.11.4 and 5					
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	Ensure that the SUT after receiving more than one INVITE sending an IAM message for each call association on receipt of a GRS message 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-I		SUT	ISUP		
	INVITE(IAM) 1	<b>→</b>	<b>→</b>	IAM		
	180 Ringing(ACM)	+	+	ACM		
	200 OK INVITE(ANM)	+	<del>(</del>	ANM		
ACK →						
	INVITE(IAM) 2	<b>→</b>	<b>→</b>	IAM		
	180 Ringing(ACM)	+	· ·	ACM		
	200 OK INVITE(ANM)	+	+	ANM		
	ACK	<b>→</b>		7 (( ( ( )		
	BYE 1	+	<del>(</del>	GRS		
	200 OK BYE	<b>→</b>	→	GRA		
	BYE 2	+				
	200 OK BYE	<b>→</b>				

TP111011	SIP reference: RFC 3261 [4]		G	ISUP reference: Q.1912.5 [1], clause 6.11.4 and 5		
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	Ensure that the SUT after receiving more than one INVITE sending an IAM message for each call association on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" 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-I		SUT		ISUP	
	INVITE(IAM) 1	<b>→</b>		<b>→</b>	IAM	
	180 Ringing(ACM)	<del>-</del>		+	ACM	
	200 OK INVITE(ANM)	<del>-</del>		+	ANM	
	ACK	<b>→</b>				
	INVITE(IAM) 2	<b>→</b>		<b>→</b>	IAM	
	180 Ringing(ACM)	<b>←</b>		+	ACM	
	200 OK INVITE(ANM)	<b>←</b>		+	ANM	
	ACK	<b>→</b>				
	BYE 1	+		+	CGB(hardware failure)	
	200 OK BYE	<b>→</b>		→	CGBA	
	BYE 2	+				
	200 OK BYE	→				

## 5.2.1.12 Receipt of the Suspend message (SUS) network initiated

TP112001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.9			
TSS reference	SIP-ISUP/Basic call/ receip "network initiated"	t of a SUSPENI	) message	with the	suspend indicator set to		
SIP selection criteria							
ISUP selection criteria							
Test purpose	"network initiated":						
SIP parameter values							
ISUP parameter values	SUS; Suspend indicator:	network initiated					
Comments	SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM) ACK	÷ + + + + + + + + + + + + + + + + + + +	SUT	<b>→ ← ←</b>	ISUP IAM ACM ANM		
	INFO(SUS) 200 OK INFO  BYE(REL) 200 OK BYE(RLC)	÷ ÷ ÷		÷	SUS(network)  REL RLC		

TP112002	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 6.9			
TSS reference	SIP-ISUP/Basic call/ receipt of "network initiated"	f a SUSF	PEND message w	ith the	suspend indicator set to		
SIP selection							
criteria							
ISUP selection	PICS 4/14						
criteria							
Test purpose	Ensure that the SUT, on receipt of a SUSPEND message with the <b>suspend indicator</b> set to "network initiated":  To is started;  after To is expired, the call is released.						
SIP parameter	INFO: encapsulated SUS						
values							
ISUP parameter	SUS; Suspend indicator: network initiated; REL: Cause value 102						
values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	ACK	<b>→</b>					
			Conversation	•			
	INFO(SUS)	+		+	SUS(network)		
	200 OK INFO	<b>→</b>			ì		
			T6 is started				
			T6 is expired				
	BYE(REL)	+	. o lo oxpirod	→	REL		
	200 OK BYE(RLC)	<b>→</b>		+	RLC		

## 5.2.1.13 Receipt of the RESume message (RES) network initiated

TP113001	SIP reference: RFC	3261 [4]			SUP reference: 12.5 [1], clause 6.10	
TSS reference	SIP-ISUP/Basic call/					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Ensure that the SUT, on receipt of a RESUME message containing the suspend/resume indicator set to "network initiated":  the RES is transferred in an INFO message.					
SIP parameter						
values						
ISUP parameter	RES; Suspend indicator: ne	twork initia	ited			
values						
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	→		→	IAM	
	180 Ringing(ACM)	+		+	ACM	
	200 OK INVITE(ANM)	+		+	ANM	
	ACK	→				
			Conversation	<u>.</u>		
	INFO(SUS)	+		+	SUS(network)	
	200 OK INFO	→				
	INFO(RES)	<b>←</b>			RES(network)	
	200 OK INFO	→				
	BYE(REL)	→		→	REL	
	200 OK BYE(RLC)	<b>←</b>		+	RLC	

#### 5.2.1.14 Receipt of Confusion message

TP114001	SIP reference: RFC 32	61 [4]		Q.19 <sup>2</sup>		P reference: [1], clause A.1.1.3	
TSS reference	ISUP-SIP/ ISUP Messages for	special o	onside	ration / Confu	usion	message	
SIP selection criteria						-	
ISUP selection criteria							
Test purpose  SIP parameter	Ensure that the SUT after receiving the INVITE with encapsulated IAM that contains an unknown parameter, sending an IAM message as received encapsulated in the INVITE request.  Ensure that when the succeeding node discards an unknown parameter and send back a Confusion message if indicated in the parameter compatibility information and the sending of a Confusion message is requested, the CFN message is transported through the SIP network encapsulated in the 183 Session Progress.  180 Ringing containing an ACM with an unknown parameter						
values	3 3 1 1 1 3 1						
ISUP parameter values	INFO with encapsulated CFN						
Comments	SIP-I					ISUP	
	INVITE	<b>→</b>			<b>→</b>	IAM	
	183 Session Progress(CFN)	+			<b>←</b>	CFN	
	180 Ringing(ACM)	+			<del>(</del>	ACM	
	200 OK INVITE(ANM)	+			<del>(</del>	ANM	
	ACK	<b>→</b>					
			Comm	nunication			
	BYE(REL)	<b>→</b>		-	<b>→</b>	REL	
	200 OK BYE(RLC)	+		·	+	RLC	

#### 5.2.1.15 Segmentation

TP115001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause A.1.1.3.1		
TSS reference	ISUP-SIP/ ISUP Message	es for spec	cial conside	eration / Red	ceip	ot of a user part test message
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Ensure that a call can be	successfu	Illy complet	ed if segme	enta	ation applies in backward
	direction.					
SIP parameter	180 Ringing - encapsulate	ed ACM: E	Backward o	all indicator	ab	sent or set to "no additional
values	information will be sent"					
	No action takes place on	the SIP si	de			
ISUP parameter	ACM: optional forward ca	ll indicator	: additiona	l information	า พ	ill be sent in a segmentation
values	message					
	SGM: optional parameters	S				
Comments	SIP-I		SU			ISUP
	INVITE(IAM)	→		-	<b>→</b>	IAM
	180 Ringing(ACM)	+		•	<b>F</b>	ACM
				•	<b>F</b>	SGM
	200 OK INVITE(ANM)	+		•	F	ANM
	ACK	<b>→</b>				
			Convers	ation		
	BYE(REL)	<b>→</b>		-	<b>&gt;</b>	REL
	200 OK BYE	+		•	F	RLC

# 5.2.2 Interworking from ISUP to SIP-I

## 5.2.2.1 Sending of the INVITE message

TP301001	SIP reference: RFC 3261	SIP reference: RFC 3261 [4] ISUP reference: Q.1912.5 [1], clause 7.1 1 a)						
TSS reference	ISUP-SIP /Basic call/Sending of th	ISUP-SIP /Basic call/Sending of the INVITE message						
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication:  • sends the INVITE message with the encapsulated IAM in the MIME body.							
SIP parameter								
values								
ISUP parameter	IAM; Called party number: with s	ending com	plete indication					
values								
Comments	ISUP/BICC		SUT	SIP-I				
	IAM -	<b>→</b>	→	INVITE(IAM)				
	ACM	F	+	180 Ringing(ACM)				
	ANM	F	+	200 OK INVITE(ANM)				
			→	ACK				
		Conv	ersation					
	REL -	<b>&gt;</b>	→	BYE(REL)				
	RLC •	F	+	200 OK BYE(RLC)				

TP301002	SIP reference: RFC 326	1 [4]		15	SUP reference:			
				2.1912	.5 [1], clause 7.1 1 b)			
TSS reference	ISUP-SIP /Basic call/Sending of the INVITE message							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT in Idle state number of digits used in the nation sends the INVITE message.		•	essage	e containing the maximum			
SIP parameter values								
ISUP parameter values	IAM; Called party number com	plete	number					
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				→	ACK			
			Conversation					
	REL	<b>→</b>		→	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP301003	SIP reference: RFC 3261	1 [4]	G		SUP reference: .5 [1], clause 7.1 1 c)			
TSS reference	ISUP-SIP /Basic call/Sending of the INVITE message							
SIP selection criteria								
ISUP selection criteria								
Test purpose	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: comp	olete numbe	er					
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	<b>←</b>		+	180 Ringing(ACM)			
	ANM	<b>←</b>		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		Con	versation					
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	<del>(</del>		+	200 OK BYE(RLC)			

TP301004	SIP reference: RFC 3261 [	<u>t]</u>			SUP reference: .5 [1], clause 7.1 1 d)		
TSS reference	ISUP-SIP /Basic call/Sending of the INVITE message						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the <b>complete called party number</b> where the end of address signalling is determined by the expiration timer T <sub>OIW1</sub> after the receipt of the latest address message:  • sends the INVITE message.						
SIP parameter values							
ISUP parameter values							
Comments	ISUP/BICC		SUT		SIP-I		
	IAM →						
		TOI	N1 expiry				
				<b>→</b>	INVITE(IAM)		
	ACM ←			+	180 Ringing(ACM)		
	ANM ←			+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
		Con	versation				
	REL →			<b>→</b>	BYE(REL)		
	RLC ←			<b>←</b>	200 OK BYE(RLC)		

TP301005	SIP reference: RFC 326	61 [4]		_	SUP reference: 2.5 [1], clause 7.1 A)				
TSS reference	ISUP-SIP/Basic call/Sending of	ISUP-SIP/Basic call/Sending of the INVITE message							
SIP selection criteria									
ISUP selection criteria	PICS 1/5								
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message with the <b>complete called party number</b> containing the Continuity Check indicator in the Nature of Connection Indicators parameter is set to indicate " <b>continuity check not required</b> ":  • sends a INVITE message.								
SIP parameter values									
ISUP parameter values									
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
		•	Conversation						
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP301006	SIP reference: RFC 3261	[4]		ISUP reference:			
	Q.1912.5 [1], clause 7.1 A)						
TSS reference	ISUP-SIP/Basic call/Sending of the	INVITE m	essage				
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5	AND NOT	PICS 4/15				
criteria							
ISUP selection criteria	PICS 1/5						
Test purpose	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 values							
Comments	ISUP		SUT	SIP-I			
	IAM =	<b>&gt;</b>					
	COT	<b>&gt;</b>	→	INVITE(IAM)			
	ACM	-	+	180 Ringing(ACM)			
	ANM	-	+	200 OK INVITE(ANM)			
			→	ACK			
		Conv	ersation				
	REL =	<b></b>	→	BYE(REL)			
	RLC		+	200 OK BYE(RLC)			

TP301007	SIP reference: RFC 3261 [4]	Į;	SUP reference:				
	Q.1912.5 [1], clause 7.1 A)						
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message						
SIP selection	NOT PICS 4/4 AND NOT PICS 4/5 AND NO	T PICS 4/15					
criteria							
ISUP selection	PICS 1/5						
criteria							
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message with the complete called party number containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check performed on previous circuit":  • sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful".						
SIP parameter values							
ISUP parameter values							
Comments	ISUP	SUT	SIP-I				
	IAM →						
	COT →	→	INVITE(IAM)				
	ACM ←	+	180 Ringing(ACM)				
	ANM ←	+	200 OK INVITE(ANM)				
		→	ACK				
	Con	versation					
	REL →	→	BYE(REL)				
	RLC ←	+	200 OK BYE(RLC)				

TP301008	SIP reference: RFC 326	1 [4]		ISUP reference:	
			Q.191	l2.5 [1], clause 7.1 A)	
TSS reference	ISUP-SIP/Basic call/Sending of				
SIP selection	NOT PICS 4/4 AND NOT PICS 4	4/5 AND NOT	PICS 4/15		
criteria					
ISUP selection	PICS 1/5				
criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit". INVITE shall not be sent if the Continuity message is received with the Continuity Indicators parameter set to "continuity check failed".				
SIP parameter					
values					
ISUP parameter					
values					
Comments	ISUP		SUT	SIP-I	
	IAM	<b>→</b>			
	COT	<b>→</b>			

TP301009	SIP reference: RFC 3261	[4]	Q.19	ISUP reference: 912.5 [1], clause 7.1 A)			
TSS reference	ISUP-SIP/Basic call/Sending of the	e INVITE me	essage				
SIP selection criteria	NOT PICS 4/4 AND NOT PICS 4/5	NOT PICS 4/4 AND NOT PICS 4/5 AND NOT PICS 4/15					
ISUP selection criteria	PICS 1/5	PICS 1/5					
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit". INVITE shall not be sent if the ISUP timer T8 expires. The SUT:  sends a REL message.						
SIP parameter values							
ISUP parameter values							
Comments	ISUP	S	SUT	SIP-I			
	IAM -	<b>→</b>		·			
		T8 (	expiry				
	REL ◀	<del>-</del>					
	RLC -	<b>→</b>					

TP301010	SIP reference: RFC 3261	4]			SUP reference:		
		Q.1912.5 [1], clause 7.1 B)					
TSS reference	ISUP-SIP/Basic call/Sending of the	INVITE m	essage				
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS	4/15					
criteria							
ISUP selection criteria	PICS 1/5 AND PICS 4/2						
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate "continuity check not required":  • sends an INVITE message without precondition using the SDP offer in the INVITE.						
SIP parameter values			<u> </u>				
ISUP parameter values							
Comments	ISUP		SUT		SIP-I		
	IAM -	•		<b>→</b>	INVITE(IAM)		
	ACM <b>←</b>	•		<b>←</b>	180 Ringing(ACM)		
	ANM +	•		<b>←</b>	200 OK INVITE(ANM)		
	→ ACK						
		Conv	ersation				
	REL -	•		<b>→</b>	BYE(REL)		
	RLC +	•		<b>←</b>	200 OK BYE(RLC)		

TP301011	SIP reference: RFC 32	61 [4]			-	SUP reference:
					<u>ર.191</u>	2.5 [1], clause 7.1 B)
TSS reference	ISUP-SIP/Basic call/Sending of			essage		
SIP selection	PICS 4/4 AND PICS 4/5 AND P	ICS 4	/15			
criteria						
ISUP selection	PICS 1/5 AND PICS 4/2					
criteria						
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate "continuity check required on this circuit":  • sends an INVITE message with precondition using the SDP offer in the INVITE. The SDP					
	Continuity message with th	e Cor	tinuity Ind	dicators pai	amet	being met is sent when the er set to "continuity check s are met in the SIP network.
SIP parameter						
values						
ISUP parameter						
values						
Comments	ISUP		S	SUT		SIP-I
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)
					+	183 Session Progress
					<b>→</b>	PRACK
					+	200 OK PRACK
	COT(successful)	<b>→</b>			<b>→</b>	UPDATE
					<b>←</b>	200 OK UPDATE
			Precond	litions met		
	ACM	+			+	180 Ringing(ACM)
	ANM	+			+	200 OK INVITE(ANM)
					<b>→</b>	ACK
			Conve	ersation		
	REL	<b>→</b>			<b>→</b>	BYE(REL)
	RLC	+			+	200 OK BYE(RLC)

TP301012	SIP reference: RFC 3261	[4]	Į;	SUP reference:	
			Q.191	2.5 [1], clause 7.1 B)	
TSS reference	ISUP-SIP/Basic call/Sending of the	INVITE m	essage		
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS	3 4/15			
criteria					
ISUP selection	PICS 1/5 AND PICS 4/2				
criteria					
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate "continuity check performed on previous circuit":  • sends an INVITE message with precondition using the SDP offer in the INVITE. The SDP offer or answer carrying the confirmation of a precondition being met is sent when the Continuity message with the Continuity Indicators parameter set to "continuity check successful" was received and the requested preconditions are met in the SIP network.				
SIP parameter	Subsection was received and	a tino roquo	otoa procoriation	s are mot in the en hetwerk.	
values					
ISUP parameter					
values					
Comments	ISUP		SUT	SIP-I	
	IAM -	<b>&gt;</b>	<b>→</b>	INVITE(IAM)	
			+	183 Session Progress	
			→	PRACK	
			+	200 OK PRACK	
	COT(successful)	<b>&gt;</b>	→	UPDATE	
			+	200 OK UPDATE	
		Precond	ditions met		
	ACM •	-	+	180 Ringing(ACM)	
	ANM	-	+	200 OK INVITE(ANM)	
			→	ACK	
		Conv	ersation		
	REL -	<b>\</b>	→	BYE(REL)	
	RLC •		+	200 OK BYE(RLC)	

TP301013	SIP reference: RFC 326	61 [4]		-	SUP reference:	
				2.191	2.5 [1], clause 7.1 B)	
TSS reference	ISUP-SIP/Basic call/Sending of	the INVITE I	nessage			
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND P	ICS 4/15				
ISUP selection criteria	PICS 1/5 AND PICS 4/2					
Test purpose	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The Continuity message is received with the Continuity Indicators parameter set to "continuity check failed". The call has been cleared before an early dialogue has been established. Ensure that the SUT:  • sends CANCEL on the SIP side.					
SIP parameter values						
ISUP parameter values						
Comments	ISUP		SUT		SIP-I	
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)	
				+	100 Trying	
	COT(unsuccessful)	<b>→</b>		<b>→</b>	CANCEL	
				+	200 OK CANCEL	
				+	487 Request Terminated	
				<b>→</b>	ACK	

TP301014	SIP reference: RFC 326	61 [4]		I;	SUP reference:
				Q.191	2.5 [1], clause 7.1 B)
TSS reference	ISUP-SIP/Basic call/Sending of	the INVIT	E message		
SIP selection	PICS 4/4 AND PICS 4/5 AND PI	ICS 4/15			
criteria					
ISUP selection	PICS 1/5 AND PICS 4/2				
criteria					
Test purpose	The SUT in Idle state, receives a				
	the Nature of Connection Indica				
	this circuit" and sends an INVI				
	INVITE. The ISUP Timer T8 exp	<b>pires</b> . The	call has bee	en cleared	d <b>before</b> an early dialogue has
	been established. Ensure that the	ne SUT:			
	<ul> <li>sends CANCEL on the SIP</li> </ul>	side.			
SIP parameter					
values					
ISUP parameter					
values					
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
				<b>←</b>	100 Trying
	T8 expires				
	REL(#47)	+		<b>→</b>	CANCEL
	RLC	<b>→</b>		<b>←</b>	200 OK CANCEL
				+	487 Request Terminated
			·	<b>→</b>	ACK

TP301015	SIP reference: RFC 3261 [4]	IS	SUP reference:			
		Q.1912	2.5 [1], clause 7.1 C)			
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE m	nessage				
SIP selection	NOT PICS 4/15					
criteria						
ISUP selection	PICS 1/4					
criteria						
Test purpose	Ensure that the SUT in Idle state, on receipt of	of an IAM message	e indicating "COT to be			
	expected":					
	<ul> <li>The sending of the INVITE is delayed un</li> </ul>					
	<ul> <li>Continuity message, with the Con</li> </ul>	tinuity Indicators p	arameter set to "continuity"			
	shall be received;					
	<ul> <li>Bearer Set-up indication - for the</li> </ul>					
	Connect Type is "notification not i	equired" was rece	ived.			
SIP parameter						
values						
ISUP parameter						
values						
Comments		SUT	SIP-I			
	IAM →					
	COT(successful) →	→	INVITE(IAM)			
	ACM ←	<b>←</b>	180 Ringing(ACM)			
	ANM ← 200 OK INVITE(ANM)					
	→ ACK					
	Con	versation				
	REL →	<b>→</b>	BYE(REL)			
	RLC ←	+	200 OK BYE(RLC)			

TP301016	SIP reference: RFC 326	1 [4]	Q.19	ISUP reference: 012.5 [1], clause 7.1 C)	
TSS reference:	ISUP-SIP/Basic call/Sending of tl	ne INVITE m		<u> </u>	
SIP selection criteria	NOT PICS 4/15		<u> </u>		
ISUP selection criteria	PICS 1/4				
SIP parameter values ISUP parameter values	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected":  The sending of the INVITE is delayed until all the following conditions are satisfied: Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received; APM with Action indicator set to "Connected" - for the forward bearer set-up cases (with, or without bearer control tunnelling) where the incoming Connect Type is "notification required", and for the fast set-up (backward) case.				
Comments	BICC		SUT	SIP-I	
	IAM	<b>→</b>			
	COT(successful)	<b>→</b>			
	APM	→	→		
	ACM	<del>-</del>	<del>(</del>	180 Ringing(ACM)	
	ANM	+	+		
			<b>→</b>	ACK	
		Con	versation		
	REL	<b>→</b>	→	BYE(REL)	
	RLC	+	+	200 OK BYE(RLC)	

TP301017	SIP reference: RFC 3261 [4]		SUP reference:				
		Q.191	2.5 [1], clause 7.1 C)				
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE	ISUP-SIP/Basic call/Sending of the INVITE message					
SIP selection	NOT PICS 4/15						
criteria							
ISUP selection	PICS 1/4						
criteria							
Test purpose	Ensure that the SUT in Idle state, on receipt	of an IAM messag	ge indicating "COT to be				
	expected":						
	<ul> <li>The sending of the INVITE delays until</li> </ul>						
	<ul> <li>Continuity message, with the Co</li> </ul>	ntinuity Indicators	parameter set to " <b>continuity</b> "				
	shall be received;						
	<ul> <li>Bearer Set-up Connect indicatio</li> </ul>	n - for the backwar	d bearer set-up case was				
	received.						
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP	SUT	SIP-I				
	IAM →						
	COT(successful) →	<b>→</b>	INVITE(IAM)				
	ACM ←	+	180 Ringing(ACM)				
	ANM ←	<b>←</b>	200 OK INVITE(ANM)				
		<b>→</b>	ACK				
	Co	nversation					
	REL →	<b>→</b>	BYE(REL)				
	RLC ←	+	200 OK BYE(RLC)				

TP301018	SIP reference: RFC 3261 [4]	_	SUP reference:			
	Q.1912.5 [1], clause 7.1 C) 2.4					
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE r	nessage				
SIP selection	NOT PICS 4/15					
criteria						
ISUP selection	PICS 1/4					
criteria						
Test purpose	Ensure that the SUT in Idle state, on receipt	of an IAM messag	e indicating "COT to be			
	expected":					
	The sending of the INVITE delays until a	III the following cor	nditions are satisfied:			
	<ul> <li>Continuity message, with the Cor</li> </ul>	tinuity Indicators	parameter set to "continuity"			
	shall be received;					
	<ul> <li>BNC set-up success indication for</li> </ul>	r cases using bear	rer control tunnelling was			
	received.	· ·	•			
SIP parameter						
values						
ISUP parameter						
values						
Comments	ISUP	SUT	SIP-I			
	IAM →					
	COT(successful) →	→	INVITE(IAM)			
	ACM ←	+	180 Ringing(ACM)			
	ANM ← 200 OK INVITE(ANM)					
	→ ACK					
	Cor	versation				
	REL →	<b>→</b>	BYE(REL)			
	RLC ←	+	200 OK BYE(RLC)			

TP301019	SIP reference: RFC 326	61 [4]		= :	SUP reference:			
TSS reference	ICUID CID/Dasia call/Canding of	Q.1912.5 [1], clause 7.1 C)						
	ISUP-SIP/Basic call/Sending of	me iiv	message					
SIP selection	NOT PICS 4/15							
criteria								
ISUP selection	PICS 1/4							
criteria								
Test purpose	expected":  sends not the INVITE if the expires:	sends <b>not</b> the INVITE if the Continuity message was not received, i.e. the BICC timer <b>T8</b>						
	<ul> <li>send REL with Caus</li> <li>of the O-IWU.</li> </ul>	e Value	e 41 ( <b>tempora</b>	ry failure)	shall be sent on the BICC side			
SIP parameter values								
ISUP parameter values								
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>						
	T8 expires							
	REL(#41) ←							
	RLC	<b>→</b>						

TP301020	SIP reference: RFC 3261 [4]			ISUP reference:			
				912.5 [1], clause 7.1 D)			
TSS reference	ISUP-SIP/Basic call/Sending of the II	NVITE me	essage				
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 4	/15					
criteria							
ISUP selection	PICS 1/4 AND PICS 4/2						
criteria							
Test purpose	The precondition signalling is concludexchange) confirmation of a precondition being resatisfied when:  Continuity message, with the Coreceived;  Bearer Set-up indication - for the	expected" sends an INVITE message with precondition using the SDP offer in the INVITE.  The precondition signalling is concluded upon sending the (within an SDP offer-answer exchange) confirmation of a precondition being met. The SDP offer or answer carrying the confirmation of a precondition being met is sent when all of the following conditions are satisfied when:  Continuity message, with the Continuity Indicators parameter set to "continuity" shall be					
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP	S	SUT	SIP-I			
	IAM →		-	INVITE(IAM)			
			•	100 <b>0</b> 000.011 10 <b>9</b> 1000			
			-	110.010			
			•	200 OK PRACK			
	COT(successful) →		-	0.2			
			•	200 OK UPDATE			
		Precond	litions met				
	ACM ←		•	10011113113(110111)			
	ANM +		•				
			-	ACK			
		Conve	ersation				
	REL →		-	(			
	RLC +		•	200 OK BYE(RLC)			

TP301021	SIP reference: RFC 3261 [4]				SUP reference:						
				1912.	5 [1], clause 7.1 D) 2.2						
TSS reference	ISUP-SIP/Basic call/Sending of the IN	IVITE me	essage								
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 4/15										
criteria											
ISUP selection	PICS 1/4 AND PICS 4/2										
criteria Test purpose	E de cal Culti III e c	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be									
	<ul> <li>expected" sends an INVITE message. The precondition signalling is conclud exchange) confirmation of a precondition satisfied when:</li> <li>Continuity message, with the Correceived;</li> <li>APM with Action indicator set to or without bearer control tunnelling.</li> </ul>	Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;									
SIP parameter	, , , , , , , , , , , , , , , , , , , ,	,									
values											
ISUP parameter											
values											
Comments	ISUP/BICC	S	UT		SIP-I						
	IAM →			<b>→</b>	INVITE(IAM)						
				4	183 Session Progress						
				1	PRACK						
				+	200 OK PRACK						
	COT(successful) →			1	UPDATE						
				+	200 OK UPDATE						
		Precond	litions met								
	ACM ←			+	180 Ringing(ACM)						
	ANM ←			+	200 OK INVITE(ANM)						
				<b>^</b>	ACK						
		Conve	ersation								
	REL →			1	BYE(REL)						
	RLC ←			+	200 OK BYE(RLC)						

TP301022	SIP reference: RFC 3261	[4]	Q.19	ISUP reference: 12.5 [1], clause 7.1 D) 2.3					
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message								
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 4/15								
criteria									
ISUP selection	PICS 1/4								
criteria									
Test purpose	The precondition signalling is concexchange) confirmation of a precondition bein satisfied when:  Continuity message, with the creceived;	Expected" sends an INVITE message with precondition using the SDP offer in the INVITE.  The precondition signalling is concluded upon sending the (within an SDP offer-answer exchange) confirmation of a precondition being met. The SDP offer or answer carrying the confirmation of a precondition being met is sent when all of the following conditions are satisfied when:  Continuity message, with the Continuity Indicators parameter set to "continuity" shall be							
SIP parameter									
values									
ISUP parameter									
values									
Comments	ISUP/BICC		SUT	SIP-I					
	IAM =	<b>&gt;</b>		→ INVITE(IAM)					
				← 183 Session Progress					
				→ PRACK					
			•	← 200 OK PRACK					
	COT(successful)	•		→ UPDATE					
				€ 200 OK UPDATE					
		Precond	ditions met						
	ACM	•		← 180 Ringing(ACM)					
	ANM			€ 200 OK INVITE(ANM)					
				→ ACK					
		Conv	ersation						
	REL -	<b>•</b>		→ BYE(REL)					
	RLC	-	•	€ 200 OK BYE(RLC)					

TP301023	SIP reference: RFC 3261 [4	1]		18	SUP reference:			
				912.	5 [1], clause 7.1 D) 2.4			
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message							
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS	4/15						
criteria								
ISUP selection	PICS 1/4 AND PICS 4/2							
criteria								
Test purpose  SIP parameter	expected" sends an INVITE messa The precondition signalling is concluexchange) confirmation of a precondition being satisfied when  Continuity message, with the Coreceived	Continuity message, with the Continuity Indicators parameter set to "continuity" shall be						
values								
ISUP parameter values								
Comments	BICC		SUT		SIP-I			
	IAM →			<b>→</b>	INVITE(IAM)			
				+	183 Session Progress			
				<b>→</b>	PRACK			
				+	200 OK PRACK			
	COT(successful) →			<b>→</b>	UPDATE			
				+	200 OK UPDATE			
		Precon	ditions met					
	ACM ←			+	180 Ringing(ACM)			
	ANM ←			+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		Conv	ersation					
	REL →			<b>→</b>	BYE(REL)			
	RLC ←			+	200 OK BYE(RLC)			

TP301024	SIP reference: RFC 3261 [4]			IS	SUP reference:		
	Q.1912.5 [1], clause 7.1 D)						
TSS reference	ISUP-SIP/Basic call/Sending of the IN	VITE me	essage				
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 4/	15					
criteria							
ISUP selection	PICS 1/4 AND PICS 4/2						
criteria							
Test purpose	The SUT in Idle state, on receipt of ar						
	in the Nature of Connection Indicators						
	sends an INVITE message with preco						
	<ul> <li>ensure that the SUT sends CANO</li> </ul>				xpires if the call has been		
	cleared <b>before</b> an early dialogue	has bee	n establishe	d.			
SIP parameter							
values							
ISUP parameter							
values	_						
Comments	BICC	S	SUT		SIP-I		
	IAM →			<b>→</b>	INVITE(IAM)		
				+	100 Trying		
		T8 e	xpires				
	REL(#47) ←			<b>→</b>	CANCEL		
	RLC →			+	200 OK CANCEL		
				+	487 Request Terminated		
				<b>→</b>	ACK		

TP301025	SIP reference: RFC 3261 [4]				SUP reference:			
	Q.1912.5 [1], clause 7.1							
TSS reference	ISUP-SIP/Basic call/Sending of the INVITE message							
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 4/15	5						
criteria								
ISUP selection	PICS 1/5 AND PICS 4/2							
criteria								
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing Continuity Check ndicator in the Nature of Connection Indicators parameter which is set to "COT to be expected". Ensure that the SUT:  sends CANCEL if on the SIP side the internal resource reservation was unsuccessful and if the call has been cleared before an early dialogue with the message has been established;  a REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU.							
SIP parameter values								
ISUP parameter values								
Comments	BICC	SU	IT		SIP-I			
	IAM →				INVITE(IAM)			
				<b>←</b>	100 Trying			
	internal resource	e reserva	ation was u	nsuc	cessful			
	REL(#47) ← CANCEL							
	RLC →				200 OK CANCEL			
					487 Request Terminated			
				<b>→</b>	ACK			

TP301026	SIP reference: RFC 32	261 [4]	Q.1	ISUP reference: Q.1912.5 [1], clause 7.1.1						
TSS reference	ISUP-SIP/Basic call/ Sending of	ISUP-SIP/Basic call/ Sending of the INVITE message								
SIP selection criteria	Based on table 6									
ISUP selection criteria										
Test purpose	Medium Requirement (TMR) contained in the IAM:  • sends an INVITE message	Ensure that the SUT in the Idle state on receipt of a IAM message, with the <b>Transmission</b> Medium Requirement (TMR) parameter set to TMR_VALUE if no USI parameter is contained in the IAM:  sends an INVITE message containing the media description defined with the "a =" "b =" and "m=" lines set to a b m LINE_VALUE.								
SIP parameter values	INVITE : a_b_m_LINE_VALUE									
ISUP parameter values	IAM: TMR : ISUP_TMR									
Comments	ISUP/BICC	SI	JT	SIP-I						
	IAM	<b>→</b>	→	INVITE(IAM)						
	ACM	+	+	180 Ringing(ACM)						
	ANM	+	+	200 OK INVITE(ANM)						
			→	ACK						
		Convei	sation							
	REL	<b>→</b>	→	BYE(REL)						
	RLC	+	<b>+</b>	200 OK BYE(RLC)						

TP301027	SIP reference: RFC 3261 [4]			(	-	SUP reference: 2.5 [1], clause 7.1.1		
TSS reference	ISUP-SIP/Basic call/ Sending	g of the	INVITE me	essage				
SIP selection criteria	Based on table 7							
ISUP selection criteria								
Test purpose	information parameter set t	Ensure that the SUT in the Idle state on receipt of an IAM message, with the user information parameter set to USI_VALUE:  • sends an INVITE message, with the media description defined with the "a = " "b =" and "m=" lines set to a b m LINE VALUE.						
SIP parameter values	INVITE: a_b_m_LINE_VALU							
ISUP parameter values	IAM: USI : ISUP_USI							
Comments	ISUP/BICC		SU	Γ		SIP-I		
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)		
	ACM	+			+	180 Ringing(ACM)		
	ANM	+			+	200 OK INVITE(ANM)		
					<b>→</b>	ACK		
			Convers	ation	•			
	REL	<b>→</b>			<b>→</b>	BYE(REL)		
	RLC	+			+	200 OK BYE(RLC)		

Table 6

	Values for test purposes TP301026									
	ISUP	SDP - a_b_m_LINE_VALUE								
	TMR parameter	m= line			b= line	a= line				
	TMR codes	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandw idth-value=""></bandw></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)				
	"speech"	Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000				
VA_02	"3,1 KHz audio"	Audio	RTP/AVP	0 and/or 8	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000				
VA_03	"64 kbit/s unrestricted"	Audio	RTP/AVP	9	AS:64	rtpmap:9 G722/8000				
	"64 kbit/s unrestricted"	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> CLEARMODE/8000</dynamic-pt>				

Table 7

	Values for test pu					rposes TP301026, TP301027						
VA		ISU	P			SDP - a_b_m_LINE_VALUE						
		USI parameter	HLC IE in ATP m= line		m= line		n= line		m= line b= line		b= line	a= line
	TMR	Information Transport Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification	<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>			
VA_01	"speech"	"Speech"	"G.711 μ-law"	Ignore	audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)			
	"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000			
VA_02	"3,1 KHz audio"	USI Absent		Ignore	audio	RTP/AVP	0 and/or 8	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000			
VA_03	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 μ-law"		audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)			
	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"		audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000			
VA_04	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64	Based on T.38.			
	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64	Based on T.38.			
VA_05	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 µ-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64	Based on T.38.			
	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 µ-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64	Based on T.38.			
VA_06	"64 kbit/s unrestricted"	"Unrestricted digital inf. W/tone/ann."	N/A	Ignore	audio	RTP/AVP	9	AS:64	Rtpmap:9 G722/8000			
VA_07	"64 kbit/s unrestricted"	"Unrestricted digital information"	N/A	Ignore	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> CLEARMODE/8000</dynamic-pt>			

TP301028	SIP reference: RFC 3	G	-	SUP reference: 2.5 [1], clause 7.1.2					
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT is mapping Called Party Number parameted to the addr-spec components.	er of the IAM:	-						
SIP parameter	INVITE: To:				Ğ				
values									
ISUP parameter									
values									
Comments	ISUP/BICC	S	SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		<b>←</b>	180 Ringing(ACM)				
	ANM	+		<b>←</b>	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
		Conve	ersation						
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		<del>(</del>	200 OK BYE(RLC)				

TP301029	SIP reference: RFC	3261 [4	]	ISUP reference: Q.1912.5 [1], clause 7.1.2						
TSS reference	ISUP-SIP/Basic call/ Sendir	ISUP-SIP/Basic call/ Sending of the INVITE message								
SIP selection criteria			_							
ISUP selection criteria										
Test purpose	<ul><li>Called Party Number param</li><li>to the addr-spec compo</li></ul>	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 values	INVITE: To: sip:; user=p				•					
ISUP parameter values										
Comments	ISUP/BICC		SUT		SIP-I					
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ACM	+		+	180 Ringing(ACM)					
	ANM	+		+	200 OK INVITE(ANM)					
				→	ACK					
			Conversation							
	REL	<b>→</b>		<b>→</b>	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP301030	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.2			
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message						
SIP selection criteria	NOT PICS 1/9						
ISUP selection criteria							
Test purpose	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-I		
	IAM	<b>→</b>					
	SAM	<b>→</b>					
	SAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
		Con	versation	-			
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP301031	SIP reference: RFC 326	1 [4]	ISUP reference: Q.1912.5 [1], clause 7.1.2				
TSS reference	ISUP-SIP/Basic call/ Sending of	the INVITE r	nessage				
SIP selection criteria	NOT PICS 1/9						
ISUP selection criteria							
Test purpose	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=phone	;					
ISUP parameter values							
Comments	ISUP/BICC	S	UT	SIP-I			
	IAM =	<b>→</b>					
	SAM	<b>&gt;</b>					
	SAM =	>	→	INVITE(IAM)			
	ACM	-	+	180 Ringing(ACM)			
	ANM	-	+	200 OK INVITE(ANM)			
			<b>→</b>	ACK			
		Conve	rsation				
	REL -	<b>→</b>	→	BYE(REL)			
	RLC •	-	<del>(</del>	200 OK BYE(RLC)			

TP301032	SIP reference: RFC 3261 [4]				ISUP reference: 12.5 [1], clause 7.1.4		
TSS reference	ISUP-SIP/Basic call/ Ser	nding of the Initial	Address M	lessage	(IAM)/		
SIP selection criteria							
ISUP selection criteria	PICS 4/3						
Test purpose		The O-IWU acting as an independent exchange shall perform the normal BICC/ISUP Hop Counter procedure as it constructs the outgoing encapsulated IAM.					
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		<b>←</b>	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
		Co	nversation				
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP301033	SIP reference: RFC 3261 [4]			Q.1	ISUP reference: 912.5 [1], clause 7.1.2		
TSS reference	ISUP-SIP/Basic call/ Sendir	ng of the IN	VITE me		<b>1</b>		
SIP selection	PICS 1/9						
criteria							
ISUP selection criteria	PICS 1/8						
SIP parameter values ISUP parameter	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, Nature of address = "International number" of the IAM to the addr-spec component of the To header field in the INVITE message.  The format of the To header field is "+CC+NDC+SN":  • the forward address information is derived from the user info component of the INVITE Request-URI.  INVITE: To:						
values							
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		<b>+</b>	200 OK INVITE(ANM)		
				→	ACK		
			Conversa	ition			
	REL	<b>→</b>		→	BYE(REL)		
	RLC	+	_	+	200 OK BYE(RLC)		

TP301034	SIP reference: RFC	3261 [4	<b>!</b> ]		_	SUP reference:  2.5 [1], clause 7.1.2
TSS reference	ISUP-SIP/Basic call/ Sendin	g of the	INVITE m	essage		
SIP selection criteria	PICS 1/9					
ISUP selection criteria	NOT PICS 1/8					
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, Nature of address = "National (significant) number" of the IAM:  • to the addr-spec component of the To header field in the INVITE message;  • the format of the To header field is "+CC+NDC+SN";  • the forward address information is derived from the user info component of the INVITE Request-URI.					
SIP parameter values	INVITE: To:					
ISUP parameter values						
Comments	ISUP/BICC		SU	Т		SIP-I
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)
	ACM	+			+	180 Ringing(ACM)
	ANM	+			+	200 OK INVITE(ANM)
				•	<b>→</b>	ACK
			Convers	ation		
	REL	<b>→</b>		•	<b>→</b>	BYE(REL)
	RLC	+			+	200 OK BYE(RLC)

TP301035	SIP reference: RFC 3261	[4]		ISUP reference: 12.5 [1], clause 7.1.2			
TSS reference	ISUP-SIP/Basic call/ Sending of the INVITE message						
SIP selection	PICS 1/9						
criteria							
ISUP selection	PICS 1/8						
criteria							
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, Nature of address = "International number" of the IAM and the and the followed SAM:  to the addr-spec component of the To header field;  the format of the To header field is "+CC+NDC+SN";  the forward address information is derived from the user info component of the INVITE Request-URI;						
SIP parameter values	INVITE: To:						
ISUP parameter values							
Comments	ISUP/BICC	SL	JT	SIP-I			
	IAM →						
	SAM →						
	SAM →		→	INVITE(IAM)			
	ACM ←		+	180 Ringing(ACM)			
	ANM ←		+	200 OK INVITE(ANM)			
			→	ACK			
		Conver	sation				
	REL →		→	BYE(REL)			
	RLC +		+	200 OK BYE(RLC)			

TP301036	SIP reference: RFC 3261 [4]			-	SUP reference:
				Q.191	12.5 [1], clause 7.1.2
TSS reference	ISUP-SIP/Basic call/ Sending of the INV	ITE m	essage		
SIP selection	PICS 1/9				
criteria					
ISUP selection	NOT PICS 1/8				
criteria					
Test purpose  SIP parameter values	Ensure that the SUT is mapping the Cal Called Party Number parameter, Nature of the IAM and the followed SAM:  to the addr-spec component of the the format of the To header field is the forward address information is a Request-URI.  INVITE: To:	of ad To hea '+CC+	Idress = ader field NDC+SN	"Natio d; l";	onal (significant) number"
ISUP parameter values					
Comments	ISUP/BICC	SU	ΙΤ		SIP-I
	IAM →				
	SAM →				
	SAM →			<b>→</b>	INVITE(IAM)
	ACM ←			+	180 Ringing(ACM)
	ANM ←			+	200 OK INVITE(ANM)
				<b>→</b>	ACK
	С	onvers	sation		
	REL →			<b>→</b>	BYE(REL)
	RLC ←			+	200 OK BYE(RLC)

## 5.2.2.2 Receipt of the SAM message after INVITE has been send

TP302001	SIP reference	RFC 3261 [4]		-	SUP reference: 912.5 [1], clause 7.2		
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after INVITE has been sent						
SIP selection criteria	PICS 3/1	-					
ISUP selection criteria							
Test purpose	Ensure if the SUT is supporting en bloc addressing towards the SIP network, subsequent SAMs received after the SUT has sent the INVITE are ignored.						
SIP parameter values							
ISUP parameter values	SAM; subsequent nu	mber (PIXIT)					
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	SAM	<b>→</b>					
	ACM	<del>(</del>		+	180 Ringing(ACM)		
	ANM	<del>(</del>		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
		•					
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP302002	SIP reference: RFC 3261 [4]			Į;	SUP reference:	
					2.5 [1], clause 7.2.1	
TSS reference	ISUP-SIP/Basic call/Receipt of SAM	after invit	e has bee	n sent		
SIP selection	PICS 3/2					
criteria						
ISUP selection	PICS 1/5					
criteria						
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to indicate "continuity check not required".  sends a INVITE message.  On receipt of a SAM from the ISUP the SUT shall:  1) Stop timer TOIW3 (if it is running);  2) TOIW2 shall be restarted and the SUT shall invoke the following procedures:  a) the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call;  b) a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent;  c) the new INVITE shall contain a new SDP offer. The O-IWU 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-I	
	IAM →			<b>→</b>	INVITE 1 (IAM)	
	SAM →			<b>→</b>	INVITE 2 (IAM)	
				+	484 Address Incomplete (1)	
				<b>→</b>	ACK	
	SAM →			<b>→</b>	INVITE 3 (IAM)	
				+	484 Address Incomplete (2)	
				<b>→</b>	ACK	
	ACM ←			+	180 Ringing (3) (ACM)	
	ANM <b>←</b>			+	200 OK INVITE (3) (ANM)	
				<b>→</b>	ACK	
		Conv	ersation			
	REL →			<b>→</b>	BYE(REL)	
	RLC 🗲			+	200 OK BYE(RLC)	

TP302003	SIP reference: RFC 326	1 [4]		ISUP reference:		
	Q.1912.5 [1], clause 7.2.1					
TSS reference	ISUP-SIP/Basic call/Receipt of S	AM after invi	te has been sent			
SIP selection	PICS 3/2 AND NOT PICS 4/15					
criteria						
ISUP selection criteria	PICS 1/5 AND PICS 4/2					
Test purpose	INVITE is sent; c) the new INVITE shall con that have already been re	connection Ir pt of the Concessful".  JP the SUT shing); I the SUT share To header find the Call-ID and the Call-ID and the call the cell within the cell within the cell second content of the cell within the cell second content of the cell within the cel	ndicators parament tinuity message whall: all invoke the followed of the new IN and From header DP offer. The O- his call. This re-us precondition attr	with the Continuity Indicators  with the Continuity Indicators  owing procedures: IVITE shall contain all digits  (including tag) as the previous  IWU may re-use any resources se of existing reserved ributes for the SDP parameters		
SIP parameter values						
ISUP parameter values						
Comments	ISUP/BICC		SUT	SIP-I		
	IAM	<b>→</b>				
	SAM	<b>→</b>				
	COT	<b>→</b>	<b>→</b>	INVITE1(IAM)		
	SAM	<b>→</b>	<b>→</b>	INVITE2(IAM)		
			+	484 Address Incomplete (1)		
			<b>→</b>	ACK		
	ACM	+	+	180 Ringing (2) (ACM)		
	ANM	<b>←</b>	+	200 OK INVITE (2) (ANM)		
			<b>→</b>	ACK		
		Conv	ersation			
	REL	<b>→</b>	<b>→</b>	BYE(REL)		
	RLC	+	+	200 OK BYE(RLC)		

TP302004	SIP reference: RFC 3261	[4]		ISUP reference:
				12.5 [1], clause 7.2.1
TSS reference	ISUP-SIP/Basic call/Receipt of SA	AM after invi	te has been sent	
SIP selection	PICS 3/2 AND NOT PICS 4/15			
criteria				
ISUP selection criteria	PICS 1/5 AND PICS 4/2			
Test purpose	INVITE is sent; c) the new INVITE shall contain that have already been re	connection In ircuit". It of the Concessful". P the SUT stang); the SUT shat To header fill; me Call-ID attain a new Served for the distributed within the	ndicators paramentinuity message whall:  all invoke the followed of the new IN and From header  DP offer. The Onis call. This re-us precondition attri-	with the Continuity Indicators  owing procedures: IVITE shall contain all digits  (including tag) as the previous  IWU may re-use any resources se of existing reserved ributes for the SDP parameters
SIP parameter values				
ISUP parameter values				
Comments	ISUP/BICC		SUT	SIP-I
	IAM	<b>→</b>		
	SAM	<b>→</b>		
	COT	<b>→</b>	<b>→</b>	INVITE 1 (IAM)
	SAM	<b>→</b>	<b>→</b>	INVITE 2 (IAM)
			+	484 Address Incomplete (1)
			<b>→</b>	ACK
	ACM	<del>(</del>	+	180 Ringing (2) (ACM)
		<del>(</del>	+	200 OK INVITE(ANM)
			<b>→</b>	ACK
		Conv	ersation	
	REL	<b>→</b>	<b>→</b>	BYE(REL)
		<del>(</del>	+	200 OK BYE(RLC)

TP302005	SIP reference: RFC 3261 [4]			ISUP reference:		
	Q.1912.5 [1], clause 7.2.1					
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after	er invite	e has been s	sent		
SIP selection	PICS 3/2 AND NOT PICS 4/15					
criteria						
ISUP selection	PICS 1/5 AND PICS 4/2					
criteria						
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" sending of INVITE is delayed.  INVITE message shall not be sent after the Continuity message was received with the Continuity Indicators parameter set to "continuity check failed".  On receipt of a SAM from the ISUP the SUT shall:  1) Stop timer TOIW3 (if it is running); 2) TOIW2 shall be restarted.					
SIP parameter values						
ISUP parameter values						
Comments	ISUP/BICC	S	UT	SIP-I		
	IAM →					
	SAM →					
	COT →					

TP302006	SIP reference: RFC 326	1 [4]	ISUP reference: Q.1912.5 [1], clause 7.2.1						
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent								
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15								
ISUP selection criteria	PICS 1/5 AND PICS 4/2								
SIP parameter values	Check indicator in the Nature of check required on this circuit" INVITE shall not be sent after the On receipt of a SAM from the ISU	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" sending of INVITE is delayed. INVITE shall not be sent after the ISUP timer T8 expires. On receipt of a SAM from the ISUP the SUT shall:  1) Stop timer TOIW3 (if it is running); 2) TOIW2 shall be restarted.							
ISUP parameter values									
Comments	ISUP/BICC		SUT	SIP-I					
	IAM	<b>→</b>							
	SAM	<b>→</b>							
			expires						
	REL	<del>(</del>							
	RLC	<b>→</b>							

TP302007	SIP refer	ence:	RFC 3261	[4]		ISUP reference: Q.1912.5 [1], clause 7.2.1		
TSS reference	ISUP-SIP/Basic	all/D	acaint of S	\	fter invit			
SIP selection	PICS 3/2 AND P	CS 4	/5 AND PIC	`S 4	15	e nas been sem		
criteria	1100 0/27/1101 1100 1/07/1100 1/10							
ISUP selection	PICS 1/5 AND PICS 4/2							
criteria	1100 1/0 /1100 4/2							
Test purpose	Check indicator i check required Sends an INVITE Indicators param preconditions are On receipt of a S 1) Stop timer TC 2) TOIW2 shall a) the Requ received b) a new IN INVITE is c) the new I that have resource in questic	nsure that the SUT in Idle state, on receipt of an IAM message containing the Continuity heck indicator in the Nature of Connection Indicators parameter which is set "continuity neck required on this circuit".  Lends an INVITE message after the reception of the Continuity message with the Continuity dicators parameter set to "continuity check successful" and after the requested reconditions are met in the SIP network.  In receipt of a SAM from the ISUP the SUT shall:  Stop timer TOIW3 (if it is running);  TOIW2 shall be restarted and the SUT shall invoke the following procedures:  a) the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call;  b) a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent;  c) the new INVITE shall contain a new SDP offer. The O-IWU 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						
SIP parameter			contains o	ligits	from the	e IAM and digits from SAM x and SAM y. The		
values	IAM is also contained							
ISUP parameter values								
Comments	10110/0100	1	0.17		loip i			
	ISUP/BICC		SUT		SIP-I	-4 (1444)		
	IAM	<b>→</b>		<b>→</b>	INVITE	E1(IAM)		
	SAM x	<b>→</b>		+	400 Ca	parion Drawnaga with out an appropriated ACM		
	СОТ	<b>→</b>		<b>~</b>	UPDA	ession Progress without encapsulated ACM		
	COT	7		<del>7</del>		(UPDATE		
	SAM y	<b>→</b>		<b>→</b>		2 (IAM and digits from SAM X + SAM Y)		
	SAIVI Y	7		<del>/</del>		Idress Incomplete (1)		
				<b>→</b>	ACK	idioso incomplete (1)		
	ACM	+		<del>′</del>		nging2 (ACM)		
	ANM	<del>`</del>		+		K INVITE(ANM)		
		<u> </u>		<u>`</u>	ACK	·····		
		С	onversatio	n -	1.0			
	REL	<b>→</b>		<b>→</b>	BYE(R	EL)		
	RLC	<b>←</b>		+		K BYE(RLC)		

TP302008	SIP refe	rence	: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause 7.2.1				
TSS reference	ISLIP-SIP/Basic	call/R	eceint of SAM a	fter i	nvite has been sent				
SIP selection criteria	PICS 3/2 AND PICS 4/5 AND PICS 4/15								
ISUP selection	PICS 1/5 AND PICS 4/2								
	1 100 1/0 / ((10 1 100 4/2								
criteria Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set or "continuity check performed on previous circuit".  Sends an INVITE message after the reception of the Continuity message with the Continuity Indicators parameter set to "continuity check successful" and after the requested preconditions are met in the SIP network.  On receipt of a SAM from the ISUP the SUT shall:  1) Stop timer TOIW3 (if it is running); 2) TOIW2 shall be restarted and the SUT shall invoke the following procedures:  a) the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call;  b) a new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent;  c) the new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved								
SIP parameter	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.  INVITE2: Request URI contains digits from the IAM and digits from SAM x and SAM y. The								
values	IAM is also conta		i contains digits	11011	The IAM and digits from SAM X and SAM y. The				
ISUP parameter									
values									
Comments:	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send an INVITE message are satisfied.								
	ISUP/BICC	<del>  _</del>	SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE1(IAM)				
	SAM x	<b>→</b>			400 Occasion Ducamaca with				
	COT	-			183 Session Progress without encapsulated ACM				
	СОТ	<b>→</b>			UPDATE				
	0444	_	+		200 OK UPDATE				
	SAM	<b>→</b>			INVITE2 (IAM and digits from SAM X + SAM Y)				
					484 Address Incomplete (1)				
	4.014	-			ACK				
	ACM	<del>(</del>	+		180 Ringing2 (ACM)				
	ANM	+			200 OK INVITE(ANM)				
				<u>→</u>	ACK				
			Conversation						
	REL	<b>→</b>			BYE(REL)				
1	RLC								

TP302009	SIP reference: RFC 3261	[4]		ISUP reference:					
				912.5 [1], clause 7.2.1					
TSS reference	ISUP-SIP/Basic call/Receipt of SA	AM after invi	te has been sen	t					
SIP selection	PICS 3/2 AND NOT PICS 4/15								
criteria	DICC 4/4 AND NOT DICC 4/0								
ISUP selection criteria	PICS 1/4 AND NOT PICS 4/2								
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".  The sending of the INVITE is delayed until all the following conditions are satisfied:								
	<ul> <li>Continuity message, with the received;</li> </ul>	Continuity I	ndicators param	neter set to "continuity" shall be					
SIP parameter	<ul> <li>Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received.</li> <li>On receipt of a SAM from the BICC the SUT shall:</li> <li>1) Stop timer TOIW3 (if it is running);</li> <li>2) TOIW2 shall be restarted and the SUT shall invoke the following procedures: <ul> <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) 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-IWU 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> </li> </ul>								
values									
ISUP parameter values									
Comments	ISUP/BICC		SUT	SIP-I					
	IAM	<b>→</b>							
	SAM x	<b>→</b>							
		<b>→</b>	→	INVITE(IAM)					
	SAM y	<b>→</b>	→	INVITE(IAM)					
	ACM	<del>(</del>	+	· · · · · · · · · · · · · · · ·					
	ANM	<del>-</del>	+						
			<b>→</b>	ACK					
		Conv	ersation						
	REL	<b>→</b>	→	\ /					
	RLC	<del>(</del>	+	200 OK BYE(RLC)					

TP302010	SIP reference: RFC 3261 [4]				SUP reference:				
				Q.191	912.5 [1], clause 7.2.1				
TSS reference	ISUP-SIP/Basic call/Receipt of SAM at	fter invite	has been	sent					
SIP selection criteria	PICS 3/2 AND NOT PICS 4/15								
ISUP selection	PICS 1/4 AND PICS 4/2								
criteria									
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".								
	The sending of the INVITE is delayed in								
	<ul> <li>Continuity message, with the Conference</li> </ul>	tinuity In	dicators pa	ırame	ter set to "continuity" shall be				
	APM with Action indicator set to "C or without bearer control tunnelling								
	required", and for the fast set-up (			g o	ormoot type is the imediation				
	On receipt of a SAM from the BICC the								
	1) Stop timer TOIW3 (if it is running);								
	2) TOIW2 shall be restarted and the S								
	a) the Request-URI and the To he	eader fie	ld of the ne	MI we	VITE shall contain all digits				
	received so far for this call;				·				
	b) a new INVITE with the same C	all-ID an	a From ne	ader (	(including tag) as the previous				
	INVITE is sent; c) the new INVITE shall contain a new SDP offer. The O-IWU 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	S	UT	<u> </u>	SIP-I				
	IAM →	ļ		<u> </u>					
	SAM x →			<u> </u>					
	COT →	ļ		<b>→</b>	INVITE(IAM)				
	SAM y	<del>                                     </del>		<b>→</b>	INVITE(IAM)				
	ACM	<u> </u>		<del>(</del>	180 Ringing(ACM)				
	ANM ←	<del>                                     </del>		<b>+</b>	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
	DEL	Conve	rsation	_	DVE(DEL)				
	REL →	<del>                                     </del>		<b>→</b>	BYE(REL)				
	RLC								

TP302011	SIP reference: RFC 3261 [4] ISUP reference:							
						12.5 [1], clause 7.2.1		
TSS reference	ISUP-SIP/Basic call/Receipt of S	SAM aft	ter invite	has been	sent			
SIP selection	PICS 3/2 AND NOT PICS 4/15							
criteria								
ISUP selection criteria	PICS 1/4 AND PICS 4/2							
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be							
rest purpose	expected".							
	The sending of the INVITE dela	vs until	all the fo	llowina co	onditio	ons are satisfied:		
						ter set to "continuity" shall be		
	received;							
	Bearer Set-up Connect indi     On receipt of a SAM from the Bl				beare	r set-up case was received.		
	Stop timer TOIW3 (if it is run		301 316	ali.				
	2) TOIW2 shall be restarted an		UT shall	invoke the	e follo	wing procedures:		
	a) the Request-URI and the							
	received so far for this c					= onan ooman an algilo		
	b) a new INVITE with the s	same Ca	all-ID and	from he	ader	(including tag) as the previous		
	INVITE is sent; c) the new INVITE shall contain a new SDP offer. The O-IWU 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							
oin .	original IAM.							
SIP parameter								
values								
ISUP parameter values								
Comments	ISUP/BICC		SL	JT		SIP-I		
	IAM	<b>→</b>						
	SAM x	→						
	COT	<b>→</b>			<b>→</b>	INVITE(IAM)		
	SAM y	<b>→</b>			<b>→</b>	INVITE(IAM)		
	ACM	+			+	180 Ringing(ACM)		
	ANM	+			+	200 OK INVITE(ANM)		
					<b>→</b>	ACK		
			Conver	sation				
	REL	<b>→</b>			<b>→</b>	BYE(REL)		
	RLC	+			+	200 OK BYE(RLC)		

TP302012	SIP reference: RFC 320	61 [4]		_	ISUP reference:				
T00 (	IOUD OID/Dania and/Danaia at	0000			12.5 [1], clause 7.2.1				
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent PICS 3/2 AND NOT PICS 4/15								
SIP selection criteria									
ISUP selection criteria	PICS 1/4 AND PICS 4/2								
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected"  The sending of the INVITE delays until all the following conditions are satisfied:  Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received;  BNC set-up success indication for cases using bearer control tunnelling was received On receipt of a SAM from the BICC the SUT shall:  Stop timer TOIW3 (if it is running);  TOIW2 shall be restarted and the SUT shall invoke the following procedures:  a) the Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call:								
	<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-IWU 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 values									
ISUP parameter values									
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	<b>→</b>							
	SAM x	<b>→</b>							
	COT	<b>→</b>		<b>→</b>	INVITE(IAM)				
	SAM y	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	<del>-</del>		+	180 Ringing(ACM)				
	ANM	<del>-</del>		+	200 OK INVITE(ANM)				
	-			<b>→</b>	ACK				
		Co	nversation	<u> </u>					
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	<del>-</del>		<del>-</del>	200 OK BYE(RLC)				

TP302013	SIP ref	erenc	e: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause 7.2.1		
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent						
SIP selection	PICS 3/2 AND PICS 4/5 AND PICS 4/15						
criteria							
ISUP selection	PICS 1/4 AND PICS 4/2						
criteria							
Test purpose	<ul> <li>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".</li> <li>Sends the INVITE message. The events:</li> <li>Continuity message, with the Continuity Indicators parameter set to "continuity" was received;</li> <li>Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received;</li> <li>are indicating the successful completion of bearer set-up.</li> <li>On receipt of a SAM from the BICC the SUT shall:</li> <li>1) Stop timer TOIW3 (if it is running);</li> <li>2) TOIW2 shall be restarted and the SUT shall invoke the following procedures: <ul> <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) 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-IWU 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> </ul> </li> </ul>						
	d) all other contents of the new INVITE are interworked from the parameters of the original IAM.						
SIP parameter				s fror	om the IAM and digits from SAM x and SAM y. The		
values	IAM is also con	taine	b				
ISUP parameter values							
Comments	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied.						
	ISUP/BICC		SUT	_	SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE1(IAM)		
	SAM x	<b>→</b>			402 Cassian Drawnas without an appropriated ACM		
	СОТ	<b>→</b>		<u>+</u>	<u> </u>		
	COT	7					
	CAMA	<b>→</b>		<u>←</u>			
	SAM y	7		7			
				1	7		
	ACM	+		<del>→</del>			
	ANM	+		+			
	CINIVI			<u> </u>			
		<b> </b>	Conversation	,	AON		
	REL	<b>→</b>	Conversation	<b>→</b>	BYE(REL)		
		<del>-</del>		<del>→</del>			
	RLC ← 200 OK BYE(RLC)						

TP302014	SIP refer	ence	: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.2.1		
TSS reference	ISUP-SIP/Basic of	all/R	eceipt of SAM	after	invite			
SIP selection	PICS 3/2 AND PI	CS 4	/5 AND PICS 4	/15				
criteria								
ISUP selection	PICS 1/4 AND PI	CS 4	1/2					
criteria								
Test purpose	Ensure that the Sexpected.	UT ii	n Idle state, on	rece	ipt of	an IAM message indicating "COT to be		
	Sends the INVITE	me	ssage. The eve	nts:				
					uity In	dicators parameter set to "continuity" was		
	APM with Acor without be	arer	control tunnellii	ng) v	vhere	ed" - for the forward bearer set-up cases (with, the incoming Connect Type is "notification"		
	required", an are indicating the		the fast set-up					
	On receipt of a S							
	1) Stop timer TC							
	2) TOIW2 shall b	e re	started and the	SUT	Γsha	Il invoke the following procedures:		
				head	der fie	eld of the new INVITE shall contain all digits		
			r for this call;					
				Call-	·ID ar	nd From header (including tag) as the previous		
	INVITE is				CF	OD offer. The O IM/I I may re use any recourses		
						OP offer. The O-IWU may re-use any resources s call. This re-use of existing reserved		
						precondition attributes for the SDP parameters		
	in questic		iii be reflected v	VICI III	i ti iC	precondition attributes for the ODF parameters		
			nts of the new	NVI	TE ar	re interworked from the parameters of the		
	original IA					- management of the control of the c		
SIP parameter			I contains digits	fror	m the	IAM and digits from SAM x and SAM y. The		
values	IAM is also conta		J			,		
ISUP parameter								
values								
Comments						nalling procedure using the SDP Offer in the		
						ed upon sending (within an SDP offer-answer		
						eing met. The SDP Offer or Answer carrying		
				ng n	net is	sent when the conditions to send a INVITE		
	message are sati	stied			1015			
	ISUP/BICC	_	SUT		SIP			
	IAM	<b>→</b>		<b>→</b>	IINV	ITE1(IAM)		
	SAM	7			400	Cassian Dragness without an appared at a ACM		
	COT	<b>→</b>		<u> </u>		Session Progress without encapsulated ACM		
	СОТ	7		<u>→</u>		DATE OK UPDATE		
	SAM	<b>→</b>						
	SAIVI	7		→ INVITE2 (IAM with digits from SAM X + SAM Y)  ← 484 Address Incomplete (1)				
				<u>▼</u>				
	ACM	+		→ ACK ← 180 Ringing2(ACM)				
	ANM	+		<del>`</del>		OK INVITE(ANM)		
	/ AI NIVI	_		<u>`</u>	ACł			
			Conversation		, (01	`		
	REL	<b>→</b>	Conversation	<b>→</b>	RYF	E(REL)		
	RLC	<del>/</del>		<del>/</del>		OK BYE(RLC)		
	INLO	~			200	ON DIE(NEO)		

TP302015	SIP refere	ence	: RFC 3261 [4]			ISUP reference:	
TSS reference	ISUP-SIP/Basic o	all/D	occipt of SAM	oftor	invito	Q.1912.5 [1], clause 7.2.1	
SIP selection	PICS 3/2 AND PI				IIIVILE	thas been sent	
criteria				713			
ISUP selection	PICS 1/4 AND PI	CS 4	/2				
criteria							
Test purpose	expected". Sends the INVITE Continuity me received: Bearer Set-u are indicating. On receipt of a So. Stop timer TO. TOIW2 shall be a) the Requereceived: b) a new INVITE is c) the new II that have resources in question.	le INVITE message. The events: tinuity message, with the Continuity Indicators parameter set to "continuity" was					
OID	original IA					LANA LES COMMENTE	
SIP parameter			i contains digits	s troi	m tne	IAM and digits from SAM x and SAM y. The	
values	IAM is also conta	nea					
ISUP parameter values							
Comments	The O IM/I I should	ا نمنه	ioto the presen	ditio	n oian	alling procedure using the SDP Offer in the	
- Commonts	INVITE. The prec exchange) the co the confirmation of message are sati	ondit nfirm of a p	ion signalling is ation of a preco recondition bei	s cor ondit	nclude tion be net is	ed upon sending (within an SDP offer-answer bing met. The SDP Offer or Answer carrying sent when the conditions to send a INVITE	
	ISUP/BICC	_	SUT	_	SIP-		
	IAM	<u>→</u>		<b>→</b>	IINVI	TE1(IAM)	
	SAM	<b>→</b>			400	Consider Dreamons without an approviated ACAA	
	COT	_		4		Session Progress without encapsulated ACM	
	СОТ	<b>→</b>		<b>→</b>	_	OK UPPATE	
	CANA			<del>/</del>		OK UPDATE	
	SAM	<b>→</b>		<u>→</u>		TE2 (IAM with digits from SAM X + SAM Y)	
						Address Incomplete (1)	
	A CM			<b>→</b>	ACK		
	ACM ANM	<u>←</u>		4	180	Ringing2(ACM) OK INVITE(ANM)	
	AINIVI	_					
			Conversation	<b>→</b>	ACK	<u> </u>	
	DEL	_	Conversation		DVE	(DEL)	
	REL	<u>→</u>		<b>→</b>		(REL)	
	RLC	<b>←</b>		+	200	OK BYE(RLC)	

TP302016	SIP refere	ence:	: RFC 3261 [4]		ISUP reference:		
TSS reference	ISLID SID/Basia c	oll/D	occipt of SAM	oftor	Q.1912.5 [1], clause 7.2.1 er invite has been sent		
SIP selection	PICS 3/2 AND PI						
criteria				/13	5		
ISUP selection	PICS 1/4 AND PI	CS 4	/2				
criteria							
Test purpose	be expected".  Sends the INVITE  Continuity management of the Invited of the Invi	ne INVITE message. The events: Itinuity message, with the Continuity Indicators parameter set to "continuity" was served; C set-up success indication for cases using bearer control tunnelling was received. Itinating the successful completion of bearer set-up, pt of a SAM from the BICC/ISUP the SUT shall: Itimer TOIW3 (if it is running); V2 shall be restarted and the SUT shall invoke the following procedures: The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call; The new INVITE with the same Call-ID and From header (including tag) as the previous NVITE is sent; The new INVITE shall contain a new SDP offer. The O-IWU 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;					
	original IA				·		
SIP parameter			I contains digits	fror	rom the IAM and digits from SAM x and SAM y. The		
values	IAM is also contain	ned					
ISUP parameter values							
Comments	Th - O 1\A/1 1 - h 1	-1 : :4	:	-1:4: -	ii		
Comments	INVITE. The prec exchange) the co the confirmation of message are satis	ondit nfirm of a p	ion signalling is ation of a preco recondition bei	s cor ondit	ion signalling procedure using the SDP Offer in the oncluded upon sending (within an SDP offer-answer dition being met. The SDP Offer or Answer carrying met is sent when the conditions to send a INVITE		
	ISUP/BICC		SUT	<b>→</b>	SIP-I		
	IAM	<u>→</u>	-	7	INVITE1(IAM)		
	SAM	<u> </u>			- 100 Coopies Progress with set as asset late 1 A ONA		
	COT			<u> </u>	J I		
	СОТ	<b>→</b>		<del>}</del>			
	CAM	_		<del>/</del>			
	SAM	<b>→</b>		<del>}</del>	, , , , , , , , , , , , , , , , , , , ,		
				<u>+</u>			
	ACM	Z-		<u>→</u>			
	ACM	+					
	ANM			<u>←</u>	\ /		
			Conversation	7	ACK		
	DEL		Conversation	_	DVE(DEL)		
	REL	<b>→</b>		<u>→</u>			
	RLC	<b>←</b>	1	+	200 OK BYE(RLC)		

TP302017	SIP reference: RFC 3261	[4]			SUP reference: 2.5 [1], clause 7.2.1			
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent							
SIP selection criteria	PICS 3/2							
ISUP selection	PICS 1/4							
criteria								
Test purpose	The SUT in Idle state, on receipt of an IAM message sends a INVITE message.  On receipt of a SAM from the BICC/ISUP the SUT shall:  1) Stop timer TOIW3 (if it is running);  2) TOIW2 shall be restarted and the SUT shall invoke the following procedures:  Ensure that if timer TOIW2 has expired, subsequent SAMs received after the SUT has sent the INVITE are ignored.							
SIP parameter values								
ISUP parameter								
values		ı						
Comments	ISUP/BICC		SUT		SIP-I			
	17 (11)	<b>→</b>			INVITE(IAM)			
	SAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
		Toiv	<sub>v2</sub> expired					
	SAM	<b>→</b>						
	ACM	<del>(</del>		+	180 Ringing(ACM)			
	ANM	<del>(</del>		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		Cor	versation					
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	<del>(</del>		+	200 OK BYE(RLC)			

TP302018	SIP reference: RFC 3261 [4]	-	SUP reference: 2.5 [1], clause 7.2.1						
TSS reference	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent								
SIP selection	PICS 3/1								
criteria									
ISUP selection	PICS 3/8								
criteria									
Test purpose	<ul> <li>The SUT in Idle state, on receipt of an IAM message.</li> <li>On receipt of a SAM from the BICC/ISUP the SUT shall:</li> <li>sends a INVITE message if the minimum number of digits for routing the call has been received in the IAM and the SAM;</li> <li>TOIW1 and TIOW2 shall be started and the SUT shall invoke the following procedures: Ensure that if timer TOIW2 has expired, subsequent SAMs received after the SUT has sent the INVITE are ignored.</li> </ul>								
SIP parameter									
values									
ISUP parameter									
values									
Comments	ISUP/BICC	SUT	SIP-I						
	IAM →								
	SAM →	→	INVITE(IAM)						
	T <sub>oiv</sub>	<sub>/2</sub> expired							
	SAM →								
	ACM ←	+	180 Ringing(ACM)						
	ANM ←	+	200 OK INVITE(ANM)						
		→	ACK						
	Cor	versation							
	REL →	<b>→</b>	BYE(REL)						
	RLC +	<b>+</b>	200 OK BYE(RLC)						

# 5.2.2.3 Sending of the ACM message

TP303001	SIP reference: RFC 3261	4]		SUP reference:				
				12.5 [1], clause 7.1,				
	Q.764 [i.12], clause 2.1.4.8							
TSS reference	ISUP-SIP /Basic call/Sending of the	e ACM mes	sage					
SIP selection	PICS 1/3							
criteria								
ISUP selection	PICS 4/9							
criteria								
Test purpose	Ensure that the SUT in Idle state, of			e containing the complete				
	called party number and the send							
	Sends the INVITE message to call			ssage with:				
	the CPS indicator set to " no it							
	the Called party's category in	ndicator se	et to "no indication	n(00)" or "ordinary subscriber				
	(01)" or "payphone (10)";							
	the interworking indicator set							
	<ul> <li>the ISUP indicator set to "ISU</li> </ul>							
	<ul> <li>the ISDN access indicator se</li> </ul>	t to "ISDN_	_ACC_IND_VAL"					
SIP parameter								
values								
ISUP parameter	IAM; Called party number: comple							
values	ACM, CPS indicator no indication		. (00)					
	Called party's category indicator	: no indicat	ion(00) or ordinal	ry subscriber (01) or payphone				
	(10)	\/AL_(DIXIT	-\					
	interworking indicator: INT_IND_		)					
	ISUP indicator: ISUP_IND_ID (PIXISDN access indicator ISDN_ACC		(DIVIT)					
Comments	ISUP/BICC		SUT	SIP-I				
Comments	IAM -		→	INVITE(IAM)				
	ACM(no indication)		7	IIIVII E(IAW)				
	CPG(Alerting)		+	180 Ringing(ACM)				
	ANM		+	200 OK INVITE(ANM)				
	ANN		<u> </u>	ACK				
		Conv	ersation	AON				
	REL -		<u> </u>	BYE(REL)				
	RLC •		<del>-</del>	200 OK BYE(RLC)				
	NLC 7		7	ZUU UN DIE(NLU)				

TP303002	SIP reference: RFC 3261	[4]		ISUP reference: .1912.5 [1], clause 7.1, 764 [i.12], clause 2.1.4.8					
TSS reference	ISUP-SIP /Basic call/ Sending of the ACM message								
SIP selection criteria	PICS 1/3								
ISUP selection criteria	PICS 4/9								
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan:  Sends the INVITE message to called user; Sends the ACM message with; the CPS indicator set to "no indication (00)"; the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)"; the interworking indicator set to "INT_IND_VAL"; the ISUP indicator set to "ISUP_IND_ID"; the ISDN access indicator set to "ISDN_ACC_IND_VAL".								
SIP parameter values	une lobit docess indicator s	ct to 10D	N_/100_IND_V	VL .					
ISUP parameter values	ACM, Backward call indicator is s indication (00)  Called party's category indicato (10) interworking indicator: INT_IND	Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: INT_IND_VAL (PIXIT) ISUP indicator: ISUP_IND_ID (PIXIT)							
Comments	ISUP/BICC		SUT	SIP-I					
	17 1171	→	•	→ INVITE(IAM)					
	- ( /	<del>-</del>							
	(	<del>-</del>		← 180 Ringing(ACM)					
	ANM	<del>-</del>		€ 200 OK INVITE(ANM)					
				→ ACK					
		Co	nversation						
	REL .	<b>&gt;</b>		→ BYE(REL)					
	RLC	<del>(</del>		€ 200 OK BYE(RLC)					

TP303003	SIP reference: RFC 3261		Q.19 Q.764	ISUP reference: 012.5 [1], clause 7.1, I [i.12], clause 2.1.4.8			
TSS reference	ISUP-SIP /Basic call/Sending of the	e ACM mes	ssage				
SIP selection criteria	PICS 1/3						
ISUP selection criteria	PICS 4/9						
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party:  • sends the INVITE message to called user;  • 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 "INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".						
SIP parameter values							
ISUP parameter values	IAM; Called party number: complete number ACM, CPS indicator no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: INT_IND_VAL (PIXIT) ISUP indicator: ISUP_IND_ID (PIXIT) ISDN access indicator ISDN ACC_IND_VAL (PIXIT)						
Comments	ISUP/BICC	5	SUT	SIP-I			
	IAM =		<b>→</b>	INVITE(IAM)			
	ACM(no indication)						
	CPG(Alerting)		+	180 Ringing(ACM)			
	ANM	•	+	200 OK INVITE(ANM)			
			<b>→</b>	ACK			
		Conv	ersation				
	REL -	·	→	BYE(REL)			
	RLC •		<b>+</b>	200 OK BYE(RLC)			

TP303004	SIP reference: RFC 3261	[4]		SUP reference:					
				, clauses 7.1 1) d), 7.3.1, 7.4					
TSS reference	ISUP-SIP /Basic call/Sending of the ACM message								
SIP selection	PICS 1/3								
criteria									
ISUP selection	NOT PICS 4/9								
criteria									
Test purpose	Ensure that the SUT in Idle state, called party number where the e timer T <sub>OIW1</sub> after the receipt of the	nd of addres	ss signalling is de						
	<ul> <li>sends the INVITE message to sends the ACM message with party's category indicator s "payphone (10)", the interwo set to "ISUP_IND_ID", the ISI</li> </ul>	n the CPS in et to "no ind rking indica	ndicator set to "n ication(00)" or "o ator set to " INT_	rdinary subscriber (01)" or IND_VAL", the <b>ISUP indicator</b>					
SIP parameter	,								
values									
ISUP parameter	IAM; Called party number: comp		-						
values	ACM, CPS indicator no indication								
	Called party's category indicato	r: no indicat	tion(00) or ordina	ry subscriber (01) or payphone					
	(10)								
	interworking indicator: INT_IND		Γ)						
	ISUP indicator: ISUP_IND_ID (PI		(51) (17)						
	ISDN access indicator ISDN_AC			Tour I					
Comments	ISUP/BICC		SUT	SIP-I					
	IAM -	<b>→</b>							
		l OIM	1 expiry						
	/ terri(iie iiiaiealieii)	<b>E</b>	<b>→</b>	INVITE(IAM)					
	G: G(::::::::::::::::::::::::::::::::::	<b>-</b>	<del>-</del>	180 Ringing(ACM)					
	ANM	E	+	200 OK INVITE(ANM)					
			<b>→</b>	ACK					
		Conv	ersation						
	: :==	<b>→</b>	<b>→</b>	BYE(REL)					
	RLC	<b>F</b>	+	200 OK BYE(RLC)					

TP303005	SIP reference: RFC 326	1 [4]		19	SUP reference:		
				Q.1912.5	5 [1], clauses 7.1, 7.3.1		
TSS reference	ISUP-SIP /Basic call/Sending of	the AC	CM message				
SIP selection	PICS 1/3						
criteria							
ISUP selection	NOT PICS 4/9						
criteria							
Test purpose	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 has been received</b> (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure):  • sends an INVITE message to the called user and after the expiration of T <sub>OIW2</sub> ;  • 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)" or "ordinary subscriber (01)" or "payphone (10)", the <b>interworking indicator</b> set to " INT_IND_VAL", the <b>ISUP indicator</b>						
	set to "ISUP_IND_ID", the IS						
SIP parameter	20.10 10012_10 , 110 10	u		2. 001.10	<u></u>		
values							
ISUP parameter	IAM; Called party number: com	plete	number				
values	ACM, CPS indicator no indication						
	Called party's category indicat	or: no	indication(00)	or ordinar	y subscriber (01) or payphone		
	(10)						
	interworking indicator: INT_INI						
	ISUP_IND_ID (F						
_	ISDN access indicator ISDN_A	CC_IN	•	)			
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	→					
	SAM	<b>→</b>					
	SAM	<b>→</b>		→	INVITE(IAM)		
			T <sub>OIW2</sub> expiry	/			
	ACM(no indication)	<b>←</b>					
	CPG(Alerting)	<b>←</b>		+	180 Ringing(ACM)		
	ANM	<b>←</b>		+	200 OK INVITE(ANM)		
				→	ACK		
			Conversation	1			
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	<b>←</b>		+	200 OK BYE(RLC)		

TP303006	SIP reference: RFC 3261 [	4]	0.404		SUP reference:				
TSS reference	Q.1912.5 [1], clauses 7.1 1) a), 7.3.1 ISUP-SIP /Basic call/Sending of the ACM message								
SIP selection criteria	PICS 1/3								
ISUP selection criteria	NOT PICS 4/9								
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number, on receipt of a 180 Ringing message.  Sends the ACM message with:  the CPS indicator set to the value in the encapsulated ACM;  the Called party's category indicator set to the value in the encapsulated ACM;  the interworking indicator set to the value in the encapsulated ACM;  the ISUP indicator set to the value in the encapsulated ACM;  the ISDN access indicator set to the value in the encapsulated ACM.								
SIP parameter values									
ISUP parameter	IAM; Called party number: comple	ete number	i						
values	ACM, Backward call indicator is set	to the valu	ie in the end	apsul	ated ACM				
Comments	ISUP/BICC		SUT		SIP-I				
	IAM 🗦			<b>→</b>	INVITE(IAM)				
	ACM <b>←</b>	•		+	180 Ringing(ACM)				
	ANM ←	•		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
		Conv	ersation						
	REL -	•		<b>→</b>	BYE(REL)				
	RLC •	•		+	200 OK BYE(RLC)				

TP303007	SIP reference: RFC 3261 [4]			IS	SUP reference:			
		12.5 [ <sup>*</sup>	1], clauses 7.1 1 a), 7.3.2					
TSS reference	ISUP-SIP /Basic call/Sending of the /	ACM mes						
SIP selection	PICS 3/1							
criteria								
ISUP selection	NOT PICS 4/9							
criteria								
Test purpose	Ensure that the SUT in Idle state, on	receipt of	an IAM me	essage	e containing the complete			
	called party number on receipt of a	183 Sess	ion Progres	ss with	n encapsulated ACM:			
	<ul> <li>sends the ACM message;</li> </ul>							
	<ul> <li>the encapsulated ACM message</li> </ul>	is sent u	nchanged b	oackw	ard.			
SIP parameter								
values								
ISUP parameter	IAM; Called party number: complete	e number						
values								
Comments	ISUP/BICC	S	UT		SIP-I			
	IAM →			<b>→</b>	INVITE(IAM)			
	ACM(no indication)			+	183 Session Progress(ACM)			
	CPG(Alerting)			4	180 Ringing(CPG)			
	ANM ← 200 OK INVITE(ANM)							
				<b>→</b>	ACK			
		Conve	rsation					
	REL →			1	BYE(REL)			
	RLC +			+	200 OK BYE(RLC)			

TP303011	SIP reference	: RFC	3261 [4]		ISUP reference:		
					Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4		
TSS reference:		ISUP-SIP /Basic call/Sending of the INVITE message					
SIP selection criteria	PICS 1/3						
ISUP selection criteria	PICS 4/2 AND NOT P	ICS 4	/9				
Test purpose	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>OIW1</sub> after the receipt of the latest address message and the continuity check is performed (ISUP) or COT is expected (BICC):  • sends the INVITE message to called user;  • the SUT shall withhold sending ACM until a successful continuity indication has been received;  • sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "INT_IND_VAL", the ISUP indicator set to " ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".						
SIP parameter values	Set to 1001_INL	<u>/_ID ,</u>	the lobit	acc	INDESS INCIDATOR SET TO TODAL ACCURACY AND		
ISUP parameter	IAM; Called party nur	nber:	complete	e nur	umber		
values	ACM,		complete				
	CPS indicator no indi	cation	(00)				
				no ind	ndication(00) or ordinary subscriber (01) or payphone		
	(10)interworking indi	cator	: INT_INI	D_VA	/AL (PIXIT)		
	ISUP indicator: ISUP	_IND_	_ID (PIXI	Γ)			
	ISDN access indicate	or ISD	N_ACC_	IND_			
Comments	ISUP/BICC		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	The state of the s		
	COT	<b>→</b>		<b>→</b>	0. 2,		
				<b></b>	200 OK UPDATE		
		$T_{C}$	<sub>DIW1</sub> expi	ry			
	ACM(no indication)	+					
	CPG(Alerting, BCi)	+		+	180 Ringing(ACM)		
	ANM	+		+			
				<b>→</b>			
		Co	nversatio	n			
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	/		

TP303012	SIP reference	: RFC	3261 [4]		ISUP reference:	
					Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4	
TSS reference	ISUP-SIP /Basic call/S					
SIP selection criteria	PICS 1/3 AND PICS 3	/2 AN	D PICS 4	/5 A	AND PICS 4/4 AND PICS 4/15	
ISUP selection criteria	PICS 4/2 AND NOT P	ICS 4	/9			
Test purpose	Encure that the SLIT if	ovorl	an addrag	ccinc	g is to be used toward the SIP network, on receipt of	
rest purpose	an IAM message containing the <b>minimum number of digits required for routing the call</b> has been received (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure) and the continuity check is performed (ISUP) or COT is expected (BICC). After the expiry of T <sub>oiw2</sub> :  • 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 "INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".					
SIP parameter	10 1001 _1110_10	, 1110	10011 400	000	indicator cot to Tobre_Neo_indb_v/te :	
values						
ISUP parameter	ACM, Backward call indicator					
values	CPS indicator r			0)		
	Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: INT_IND_VAL (PIXIT) ISUP indicator: ISUP_IND_ID (PIXIT) ISDN access indicator ISDN_ACC_IND_VAL (PIXIT)  CPG: Event indicator = ALRTING and the BCI from the ACM encapsulated in the received 180 Ringing					
Comments	ISUP/BICC		SUT		SIP-I	
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)	
				+	183 Session Progress without encapsulated ACM	
	COT	<b>→</b>		<b>→</b>	UPDATE	
				+	200 OK UPDATE	
		T <sub>C</sub>	<sub>DIW2</sub> expi	ry		
	ACM(no indication)	+				
	CPG(Alerting, BCi)	+		+	180 Ringing(ACM)	
	ANM	+		<b>←</b>	200 OK INVITE(ANM)	
				<b>→</b>	ACK	
		Co	nversatio	n		
	REL	<b>→</b>		<b>→</b>	BYE(REL)	
	RLC	+		+	200 OK BYE(RLC)	

TP303013	SIP refere	ence	RFC 3261 [4]		ISUP reference:		
					Q.1912.5 [1], clauses 7.1, 7.3.1, 7.4		
TSS reference	ISUP-SIP /Basic call/Sending of the ACM message						
SIP selection	PICS 1/3						
criteria							
ISUP selection	PICS 4/2 AND NO	OT P	CS 4/9				
criteria							
Test purpose					pt of an IAM message containing the complete		
					k is performed (ISUP) or COT is expected (BICC)		
	indication receipt			essa	ge:		
			nessage with:				
					in the encapsulated ACM;		
					ator set to the value in the encapsulated ACM;		
					the value in the encapsulated ACM;		
					e in the encapsulated ACM;		
CID novemeter	o the ISDN a	the ISDN access indicator set to the value in the encapsulated ACM.					
SIP parameter							
values ISUP parameter	IAM; Called party	, n	nhar: complete	, nun	phor		
values					value in the encapsulated ACM		
Comments	ISUP/BICC	all II	SUT	ıne	SIP-I		
Comments	IAM	<b>→</b>	301	<b>→</b>	INVITE(IAM)		
	IAIVI			<del>/</del>	183 Session Progress without encapsulated ACM		
	СОТ	<b>→</b>		<u>→</u>	UPDATE		
	COT			<del>/</del>	200 OK UPDATE		
	ACM	<del>-</del>		<del>-</del>	180 Ringing(ACM)		
	ANM	<del>-</del>		<del>-</del>	200 OK INVITE(ANM)		
	ALVIVI			$\rightarrow$	ACK		
			Conversation		NOIL		
	REL	<b>→</b>	Conversation	<b>→</b>	BYE(REL)		
	RLC	<del>′</del>		<del>-</del>	200 OK BYE(RLC)		
L	ILLO				200 ON DIE(NEO)		

TP303014	SIP reference: RFC 326	1 [4]			SUP reference:			
				12.5 [	1], clauses 7.1, 7.3.1, 7.4			
TSS reference	ISUP-SIP /Basic call/Sending of	the INVITI	message					
SIP selection criteria	PICS 1/3 AND PICS 3/2 AND NO	OT PICS 4	/15					
ISUP selection	PICS 3/8 AND PICS 4/2 AND NO	OT PICS 4	/9					
criteria								
Test purpose	Ensure that the SUT if <b>overlap addressing is to be used toward the SIP network</b> , on receipt of an IAM message containing the <b>minimum number of digits required for routing the call</b> has been received (start timer T <sub>OIW2</sub> and invoke the appropriate outgoing SIP signalling procedure) and the continuity check is performed (ISUP) or COT is expected (BICC). After the expiry of T <sub>Oiw2</sub> :							
	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 "INT_IND_VAL", the ISUP indicator set to "ISUP_IND_ID", the ISDN access indicator set to "ISDN_ACC_IND_VAL".							
SIP parameter	,	10 100 , 110 100 1 00000 110 100 100 100 100 1						
values								
ISUP parameter	ACM, Backward call indicator	ACM, Backward call indicator						
values	CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: INT_IND_VAL (PIXIT) ISUP indicator: ISUP_IND_ID (PIXIT) ISDN access indicator: ISDN_ACC_IND_VAL (PIXIT) CPG: Event indicator = ALRTING and the BCI from the ACM encapsulated in the received							
	180 Ringing							
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	<b>→</b>						
	COT	<b>→</b>		<b>→</b>	INVITE(IAM)			
		T	<sub>DIW2</sub> expiry					
	ACM(no indication)	+						
	CPG(Alerting)	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		C	onversation	•				
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP303015	SIP reference: RFC 3261 [4]			SUP reference:				
			Q.1912.5 [ <sup>*</sup>	1], clauses 7.1, 7.3.1; 7.4				
TSS reference	ISUP-SIP /Basic call/Sending of the ACM message							
SIP selection	PICS 1/3 AND NOT PICS 4/15							
criteria								
ISUP selection	PICS 4/2 AND NOT PICS 4/9							
criteria								
Test purpose	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message:  Sends the ACM message with:  the CPS indicator set to the value in the encapsulated ACM;  the Called party's category indicator set to the value in the encapsulated ACM;  the interworking indicator set to the value in the encapsulated ACM;  the ISUP indicator set to the value in the encapsulated ACM;  the ISDN access indicator set to the value in the encapsulated ACM.							
SIP parameter values								
ISUP parameter	IAM; Called party number: complete no	umber						
values	ACM, Backward call indicator is set to the	ne value in the	e encapsu	ated ACM				
Comments	ISUP/BICC	SUT		SIP-I				
	IAM →							
	COT →		→	INVITE(IAM)				
	ACM ←		←	180 Ringing(ACM)				
	ANM ←		←	200 OK INVITE(ANM)				
			→	ACK				
		Conversation	n					
	REL →		→	BYE(REL)				
	RLC +		<b>←</b>	200 OK BYE(RLC)				

# 5.2.2.4 Sending of the CPG message

TP304001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 7.1, 7.3.1					
TSS reference	ISUP-SIP /Basic call/ Send	ISUP-SIP /Basic call/ Sending of the CPG message							
SIP selection	PICS 3/1								
criteria									
ISUP selection	PICS 3/8	PICS 3/8							
criteria									
Test purpose	Ensure that the SUT, having sent a ACM message with called party status "no indication" on receipt of a 180 Ringing with a encapsulated ISUP message:  • sends the CPG message with the <b>event indicator</b> set to "Alerting".								
SIP parameter									
values									
ISUP parameter	ACM: BCi called party status indicator = no indication								
values	CPG: Event Indicator = ALI	ERTING, BCi	as received from	om the e	encapsulated ACM				
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	→							
	SAM	<b>→</b>							
	SAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
		T <sub>OIW2</sub> expiry							
	ACM(no indication)	+							
	CPG(Alerting BCi)	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversation						
	REL	<b>→</b>		→	BYE(REL)				
	RLC	←		<b>←</b>	200 OK BYE(RLC)				

TP304002	SIP reference: RFC 32	61 [4]			SUP reference:				
	Q.1912.5 [1], clauses 7.1, 7.3.1								
TSS reference	ISUP-SIP /Basic call/ Sending c	ISUP-SIP /Basic call/ Sending of the CPG message							
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT, having sent a ACM message with called party status "no indication" on receipt of a 183 Session progress message with a encapsulated ISUP message:  • sends the CPG message with the <b>event indicator</b> set to "Alerting".								
SIP parameter values									
ISUP parameter									
values									
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	→		<b>→</b>	INVITE(IAM)				
	ACM(no indication)	+		+	183 Session Progress(ACM)				
	CPG(Alerting)	+		+	180 Ringing(CPG)				
	ANM	+		+	200 OK INVITE(ANM)				
	→ ACK								
		С	onversation						
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

## 5.2.2.5 Sending of the ANM message

TP305001	SIP reference: RFC 3261 [	4]	ISUP reference: Q.1912.5 [1], clause 7.5						
TSS reference	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	call, it shall stop timer TOIW2 (if rur • send ANM as determined by B	Ensure that the SUT having sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer TOIW2 (if running):  send ANM as determined by BICC/ISUP procedures;  stop any existing awaiting answer indication (e.g. ringing tone).							
SIP parameter	200 OK INVITE;		o (o.ggg	103).					
values									
ISUP parameter	ANM;								
values									
Comments	ISUP/BICC	SU	Т	SIP-I					
	IAM →		<b>→</b>	INVITE(IAM)					
	ACM ←		+	180 Ringing(ACM)					
	ANM ←		+	200 OK INVITE(ANM)					
	→ ACK								
	Conversation								
	REL →		→	BYE(REL)					
	RLC <del>C</del>		+	200 OK BYE(RLC)					

## 5.2.2.6 Sending of the CON message

TP306001	SIP reference: RFC 3261 [4]			ISUP reference:				
			Q.1912	.5 [1], clauses 7.5, 7.5.1				
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT, having not sent th		nessage, on	receipt of a 200 OK INVITE for				
	this call, it shall stop timer TOIW2 (if run	٠,						
	<ul> <li>send CON as determined by BICC/</li> </ul>							
	Stop any existing awaiting answer indica	ation (e.	g. ringing ton	e) BCI encoded as received in				
	the encapsulated CON.							
SIP parameter	200 OK INVITE;							
values								
ISUP parameter	CON; interworking indicator: INT_IND	)_VAL (F	PIXIT)					
values	ISUP indicator: ISUP_IND_ID (PIXIT)							
	ISDN access indicator ISDN_ACC_INI	$D_{VAL}$ (	PIXIT)					
	CPS indicator: no indication							
Comments	ISUP/BICC	SUT		SIP-I				
	IAM →		→	INVITE(IAM)				
	CON ←		<del>-</del>	200 OK INVITE(CON)				
	→ ACK							
		Conversa	ation					
	REL →		→	BYE(REL)				
	RLC ←		+	200 OK BYE(RLC)				

## 5.2.2.7 Receipt of the Release message (REL)

TP307001	SIP reference: RFC 3	FC 3261 [4] ISUP reference:						
			Q.1912	2.5 [1], clause 7.7.1, 1)				
TSS reference	ISUP-SIP/Basic call/ Receipt of	of the Release	nessage (REL)	/				
SIP selection		*						
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT after receiving the IAM but before an INVITE has been sent. On receipt of a REL message: no action is required on the SIP side other than to terminate local procedures if any are in progress.							
SIP parameter								
values								
ISUP parameter								
values								
Comments	ISUP/BICC	SI	JT	SIP-I				
	IAM	<b>→</b>						
	REL	<b>→</b>						
	RLC	+						

TP307002	SIP reference: RFC 32	61 [4]		SUP reference:				
TSS reference	Q.1912.5 [1], clause 7.7.1 2)							
SIP selection	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/							
criteria								
ISUP selection criteria								
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message <b>before</b> any response message has been received which establishes a confirmed dialogue:  the SUT shall hold the REL message until a SIP response has been received;  the SUT shall send a BYE request.							
SIP parameter values								
ISUP parameter values								
Comments	ISUP/BICC	S	SUT	SIP-I				
	IAM	<b>→</b>	<b>→</b>	INVITE(IAM)				
	REL	<b>→</b>						
	RLC	+						
			+	200 OK INVITE(CON)				
			<b>→</b>	ACK				
			<b>→</b>	BYE(REL)				
			+	200 OK BYE(RLC)				

TP307003	SIP reference: RFC 32	61 [4]	_	SUP reference: 5 [1], clause 7.7.1 2) 3)					
TSS reference	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose  SIP parameter	<ul> <li>Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message before a 200 OK SIP response message has been received:</li> <li>the SUT shall hold the REL message. A CANCEL is sent when any SIP response was 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> </ul>								
values									
ISUP parameter									
values									
Comments	ISUP/BICC	SU	T	SIP-I					
	IAM	<b>→</b>	→	INVITE(IAM)					
			+	100 Trying					
	REL	→							
	RLC	<b>←</b>	<b>→</b>	CANCEL(REL)					
			+	200 OK INVITE(CON)					
			→	ACK					
			+	200 OK CANCEL					
			→	BYE(REL)					
			+	200 OK BYE(RLC)					

TP307004	SIP reference: RFC 32	61 [4]		ISUP reference:				
TSS reference	Q.1912.5 [1], clause 7.7.1 2) 3) ISUP-SIP/Basic call/ Receipt of the Release message (REL)/							
SIP selection criteria	Tool on / Baole sail/ (toosipt of	110 11010000	moodago (rt <u>zz)</u>					
ISUP selection criteria								
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message before an early dialogue with the message 100 Trying has been established:  the SUT shall hold the REL message until a 100 Trying response has been received; the SUT shall send a CANCEL The received REL is encapsulated in the CANCEL.							
SIP parameter values								
ISUP parameter values								
Comments	ISUP/BICC IAM REL RLC	÷ ÷ ÷ ÷	######################################	SIP-I INVITE(IAM)  100 Trying CANCEL(REL) 200 OK CANCEL 487 Request terminated ACK				

TP307005	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.7.1 4)						
TSS reference	ISUP-SIP/Basic call/ Receipt c	f the R	elease me	ssage (REL)/						
SIP selection										
criteria										
ISUP selection criteria										
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message <b>after</b> a 200 OK response message has been received:  the SUT shall hold the REL message until an ACK has been sent;  the SUT shall send a BYE request. The received REL is encapsulated in the BYE.									
SIP parameter values		-			•					
ISUP parameter values										
Comments	ISUP/BICC		SUT	•	SIP-I					
	IAM	<b>→</b>		→	INVITE(IAM)					
	ACM	+		+	180 Ringing(ACM)					
	ANM	+		+	200 OK INVITE(ANM)					
	REL	<b>→</b>								
	RLC	+		<b>→</b>	ACK					
				<b>→</b>	BYE(REL)					
				<b>+</b>	200 OK BYE(RLC)					

TP307006	SIP reference: RFC 3	261 [4]	ISUP reference: Q.1912.5 [1], clause 7.7.1 3)					
TSS reference	ISUP-SIP/Basic call/ Receipt of	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/						
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	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 or BYE request. The received REL is encapsulated in the BYE or CANCEL.							
SIP parameter values								
ISUP parameter values								
Comments	ISUP/BICC	SU	Т	SIP-I				
	IAM	<b>→</b>	<b>→</b>	INVITE(IAM)				
			+	SIP_MESSAGE_VA				
	REL	<b>→</b>						
	RLC	<b>←</b>						
	CASE A							
			→	CANCEL(REL)				
			+	200 OK CANCEL				
			+	487 Request terminated				
			<b>→</b>	ACK				
	CASE B							
			<b>→</b>	BYE(REL)				
			+	200 OK BYE				
			+	487 Request terminated				
			<b>→</b>	ACK				

Table 8

	Values for test purpose TP307106					
VA SIP MESSAGE_VA						
VA_1	VA_1 180 Ringing					
VA_2	VA_2 181 Call Is Being Forwarded					
VA_3	VA_3 182 Queued					
VA_4	183 Session Progress					

# 5.2.2.8 Sending of a REL message (REL) / receipt of a backward BYE

TP308001	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause 7.7.2					
TSS reference	ISUP-SIP /Basic call/ Sending of the Release message (REL)/							
SIP								
selection								
criteria								
ISUP								
selection								
criteria								
Test	Ensure that the SUT after receiving the	he IAM s	ends out an	INVITE m	essage and on receipt of a			
purpose	BYE message in the confirmed dialog	gue:			-			
	<ul> <li>sends a REL message construct</li> </ul>	ed from	the encapsul	ated REL	in the received BYE.			
SIP								
parameter								
values								
ISUP	REL; Cause value "Normal call cleari	ng"						
parameter		-						
values								
Comments	ISUP/BICC	SU			SIP-I			
	IAM =	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	F	•	<del>(</del>	180 Ringing(ACM)			
	ANM	<b>F</b>		<del>(</del>	200 OK INVITE(ANM)			
			ŀ	<b>→</b>	ACK			
	C	Conversa	ition					
		F		<del>(</del>	BYE(REL)			
	RLC -	<b>→</b>	j.	<del>)</del>	200 OK BYE(RLC)			

TP308002	SIP reference: RFC 3	261 [4	]	(		SUP reference: 2.5 [1], clause 7.7.6		
TSS reference	ISUP-SIP /Basic call/ Sending of the Release message (REL)/							
SIP selection criteria					,			
ISUP selection criteria								
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA:  sends a REL message constructed from the encapsulated REL.							
SIP parameter values				•				
ISUP parameter values	REL; cause value: CV_ISUP							
Comments	ISUP/BICC		SU	T		SIP-I		
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)		
					+	100 Trying		
	REL	+			+	SIP_Failure_VA(REL)		
	RLC	<b>→</b>			<b>→</b>	ACK		

Table 9

Values for test purpose TP308002						
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	402 Payment Required				
VA_03	127 Interworking	403 Forbidden				
VA_04	1 Unallocated number	404 Not Found				
VA_05	127 Interworking	405 Method Not Allowed				
VA_06	127 Interworking	406 Not Acceptable				
VA_07	127 Interworking	408 Request Timeout				
VA_08	22 Number changed (without diagnostic)	410 Gone				
VA_9	127 Interworking	423 Interval Too Brief				
VA_10	20 Subscriber absent	480 Temporarily Unavailable				
VA_11	127 Interworking	481 Call/Transaction does not exist				
VA_12	127 Interworking	482 Loop Detected				
VA_13	127 Interworking	483 Too many hops				
VA_14	127 Interworking	485 Ambiguous				
VA_15	17 User busy	486 Busy Here				
VA_16	127 Interworking	488 Not acceptable here				
VA_17	127 Interworking	493 Undecipherable				
VA_18	127 Interworking	500 Server Internal error				
VA_19	127 Interworking	501 Not implemented				
VA_20	127 Interworking	502 Bad Gateway				
VA_21	127 Interworking	504 Server timeout				
VA_22	17 User busy	600 Busy Everywhere				
VA_23	21 Call rejected	603 Decline				
VA_24	1 Unallocated number	604 Does not exist anywhere				
VA_25	127 Interworking	606 Not acceptable				

TP308003	SIP reference: RFC	ISUP reference: Q.1912.5 [1], clause 7.7.6						
TSS reference	ISUP-SIP /Basic call/ Sending of the Release message (REL)/							
SIP selection criteria	NOT PICS 4/10							
ISUP selection criteria								
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message, on receipt of a Failure message 487 Request terminated:  no action is taken on the ISUP if a CANCEL request was previously sent before an answer to an INVITE was received.							
SIP parameter values								
ISUP parameter values								
Comments	ISUP/BICC		SU	Т		SIP-I		
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)		
					+	100 Trying		
	REL	<b>→</b>			<b>→</b>	CANCEL(REL)		
	RLC	+			+	200 OK CANCEL		
					<b>←</b>	487 Request Terminated		
					<b>→</b>	ACK		

TP308004	SIP reference: RFC 3261 [4] ISUP reference: Q.1912.5 [1], clause 7.7.								
TSS reference	ISUP-SIP /Basic call/ Sending of the Release message (REL)/								
SIP selection criteria	Joseph Garage	<u> </u>							
ISUP selection criteria									
Test purpose	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:  sends a REL message constructed from the encapsulated REL.								
SIP parameter values				•					
ISUP parameter values	REL; cause value: CV_ISUP	REL; cause value: CV_ISUP							
Comments	ISUP/BICC		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	XXX	+		+	SIP MESSAGE_VA				
	REL	+	_	+	SIP_Failure_VA(REL)				
	RLC		<b>→</b>	ACK					

#### Table 10

	Values for test purpose TP308004					
VA SIP MESSAGE_VA						
VA_1 180 Ringing						
VA_2	183 Session Progress					

Table 11

Values for test purposes TP308004						
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	402 Payment Required				
VA_03	127 Interworking	403 Forbidden				
VA_04	1 Unallocated number	404 Not Found				
VA_05	127 Interworking	405 Method Not Allowed				
VA_06	127 Interworking	406 Not Acceptable				
VA_07	127 Interworking	408 Request Timeout				
VA_08	22 Number changed (without diagnostic)	410 Gone				
VA_09	127 Interworking	423 Interval Too Brief				
VA_10	20 Subscriber absent	480 Temporarily Unavailable				
VA_11	127 Interworking	481 Call/Transaction does not exist				
VA_12	127 Interworking	482 Loop Detected				
VA_13	127 Interworking	483 Too many hops				
VA_14	127 Interworking	485 Ambiguous				
VA_15	17 User busy	486 Busy Here				
VA_16	127 Interworking	488 Not acceptable here				
VA_17	127 Interworking	493 Undecipherable				
VA_18	127 Interworking	500 Server Internal error				
VA_19	127 Interworking	501 Not implemented				
VA_20	127 Interworking	502 Bad Gateway				
VA_21	127 Interworking	504 Server timeout				
VA_22	17 User busy	600 Busy Everywhere				
VA_23	21 Call rejected	603 Decline				
VA_24	1 Unallocated number	604 Does not exist anywhere				
VA_25	127 Interworking	606 Not acceptable				

TP308005	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.7.6					
TSS reference	ISUP-SIP /Basic call/ Sending	ISUP-SIP /Basic call/ Sending of the Release message (REL)/							
SIP selection criteria	NOT PICS 4/10	NOT PICS 4/10							
ISUP selection criteria									
Test purpose	Ensure that the SUT after receiving the IAM sends out an INVITE message a 180 ringing message has been received on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA:  • sends a REL message constructed from the encapsulated REL.								
SIP parameter values				•					
ISUP parameter values	REL; cause value: CV_ISUP								
Comments	ISUP/BICC		SU	Γ		SIP-I			
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ACM	+			<del>(</del>	180 Ringing			
	REL	+			<del>(</del>	SIP_Failure_VA(REL)			
	RLC	<b>→</b>			<b>→</b>	ACK			

Table 12

Values for test purposes TP308005						
VA	←REL (Cause Value) CV_ ISUP	←4XX/5XX/6XX SIP message SIP_Failure_VA				
VA_01	127 Interworking	408 Request timeout				
VA_02	17 User busy	486 Busy Here				
VA_03	17 User busy	600 Busy Everywhere				
VA_04	21 Call rejected	603 Decline				

TP30806	SIP reference: RFC 3	3261 [4	1]	ISUP reference: Q.1912.5 [1], clause 7.7.6				
TSS reference	ISUP-SIP /Basic call/ Sending of the Release message (REL)/							
SIP selection criteria	NOT PICS 4/21							
ISUP selection criteria								
Test purpose	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 CV_ISUP.							
SIP parameter values								
ISUP parameter values	REL; cause value: CV_ISUP							
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
				+	100 Trying			
	REL	+		+	SIP_Response_VA			
	RLC	<b>→</b>		<b>→</b>	ACK			

Table 13

	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					

Mapping of Cause Indicators parameter into SIP Reason header fields.

Table 14

Cause indications parameter field	Value of parameter field	component of SIP Reason header field	Component value
-	-	Protocol	"Q.850"
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 Q.850 (see note 2)

NOTE 1: "XX" is the Cause Value as defined in ITU-T Recommendation Q.850 [3].

NOTE 2: Due to the fact that the Cause Indications parameter does not include the definition text as defined in table 1/ITU-T Recommendation Q.850 [3] this is based on provisioning in the O-IWU.

#### 5.2.2.9 Autonomous release at O-IWU

#### 5.2.2.9.1 Receipt of Reset Circuit message (RSC)

TP309001	SIP reference: RFC 32	61 [4]		SUP reference:				
			Q.1912.5 [1],	clauses 7.7.1, 1), 7.7.4, 7.7.5				
TSS reference	ISUP-SIP/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	Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of a RSC message:  • no action is required on the SIP side other than to terminate local procedures if any are in progress.							
SIP parameter values								
ISUP parameter values								
Comments	ISUP/BICC	SU	IT	SIP-I				
	IAM	<b>→</b>						
	RSC	<b>→</b>						
	RLC	<b>←</b>						

TP309002	SIP reference: RFC 3	261 [4]		-	SUP reference: ], clauses 7.7.1, 7.7.4, 7.7.5			
TSS reference	ISUP-SIP/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	sending a INVITE message or response message has been to the SUT shall hold the RS	the Color and the tree message driving of the second secon						
SIP parameter values	CANCEL or BYE: A REL is en							
ISUP parameter values								
Comments	ISUP/BICC		SU	Т	SIP-I			
	IAM	<b>→</b>		→	INVITE(IAM)			
	RSC	<b>→</b>						
	RLC	+						
				+	SIP_MESSAGE_VA			
	CASE A							
				→	CANCEL			
				+	200 OK CANCEL			
				+	487 Request terminated			
				<b>→</b>	ACK			
	CASE B							
				→	BYE(REL#31)			
				+	200 OK BYE(RLC)			
				+	487 Request terminated			
			_	→	ACK			

Table 15

Values for test purpose TP309002						
VA	SIP MESSAGE_VA					
VA_1	100 Trying					
VA_2	180 Ringing					
VA_3	183 Session Progress					

TP309003	SIP reference: RFC 326	SIP reference: RFC 3261 [4]		SUP reference:				
			Q.1912.5 [1]	, clauses 7.7.1, 7.7.4, 7.7.5				
TSS reference	ISUP-SIP/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	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message before a 200 OK response message has been received:  on subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent The RSC is not encapsulated.							
SIP parameter values	BYE: A REL is encapsulated with	cause 31						
ISUP parameter values								
Comments	ISUP/BICC	SU	ΙΤ	SIP-I				
	IAM = \frac{1}{2}	•	<b>→</b>	INVITE(IAM)				
	RSC <del>1</del>	<b>&gt;</b>						
	RLC •	-						
	← 200 OK INVITE(CON)							
			→	ACK				
			→	BYE(REL#31)				
			+	200 OK BYE(RLC)				

TP309005	SIP reference: RFC 3	261 [4	ISUP reference:						
	Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5								
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message								
	(GRS) or Circuit group blocking	ng mes	sage (CGI	<ol><li>with the indi-</li></ol>	cation hardware failure				
	oriented								
SIP selection criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT after rec								
	sending a INVITE message w								
	message on receipt RSC mes	_		•	•				
	<ul> <li>the SUT shall send a BYE</li> </ul>			SC is not encap	sulated.				
SIP parameter	BYE: A REL is encapsulated v	with ca	use 31						
values									
ISUP parameter									
values									
Comments	ISUP/BICC		SU	Т	SIP-I				
	IAM	<b>→</b>		→	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
	RSC	<b>→</b>		→	BYE(REL#31)				
	RLC	<b>←</b>		+	200 OK BYE(RLC)				

TP309006	SIP reference: RFC 3	261 [4]			ISUP reference:		
					], clauses 7.7.1, 7.7.4, 7.7.5		
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure						
		ng mess	age (CGI	3) with the ind	ication hardware failure		
OID and and an	oriented						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established.  The SUT shall send a CANCEL or BYE request The RSC is not encapsulated.						
SIP parameter values	CANCEL or BYE: A REL is er	ncapsula	ited with	cause 31			
ISUP parameter							
values							
Comments	ISUP/BICC		SU		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	SIP_MESSAGE_VA		
	RSC	<b>→</b>					
	RLC	<b>←</b>					
	CASE A						
				<b>→</b>	CANCEL		
				+	200 OK CANCEL		
				+	487 Request terminated		
				→	ACK		
	CASE B						
				<b>→</b>	BYE(REL#31)		
				+	200 OK BYE(RLC)		
				+	487 Request terminated		
				<b>→</b>	ACK		

Table 16

	Values for test purpose; TP309006						
VA SIP MESSAGE_VA							
VA_1	180 Ringing						
VA 2	183 Session Progress						

## 5.2.2.9.2 Receipt of Circuit group reset message (GRS)

TP309007	SIP reference: RFC 3261	[4]	Q.191	ISUP reference: 2.5 [1], clauses 7.7.1, 1), 7.7.4, 7.7.5				
TSS reference		P/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message r Circuit group blocking message (CGB) with the indication hardware failure						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of GRS message:  no action is required on the SIP side other than to terminate local procedures if any are in progress.							
SIP parameter values								
ISUP parameter values								
Comments	ISUP/BICC IAM •• GRS		Т	SIP-I				
	GRS GRA							

TP309008	SIP reference: RFC 3	261 [4]			ISUP reference:		
				Q.1912.5 [1	], clauses 7.7.1, 7.7.4, 7.7.5		
TSS reference		ISUP-SIP/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	sending a INVITE message or response message has been  the SUT shall hold the GF	the Cort of the first the Creation of the Cort of the					
SIP parameter values							
ISUP parameter values							
Comments	ISUP/BICC		SU	Т	SIP-I		
	IAM	<b>→</b>		→	INVITE(IAM)		
	GRS	<b>→</b>					
	GRA	+					
				+	SIP_MESSAGE_VA		
	CASE A						
				→	CANCEL		
				+	200 OK CANCEL		
				+	487 Request terminated		
				→	ACK		
	CASE B	1					
				<b>→</b>	BYE(REL#31)		
				<del>-</del>	200 OK BYE(RLC)		
				+	487 Request terminated		
				<b>→</b>	ACK		

Table 17

	Values for test purpose TP309008					
VA	SIP MESSAGE_VA					
VA_1	100 Trying					
VA_2	180 Ringing					
VA_3	183 Session Progress					

TP309009	SIP reference: RFC 3	261 [4]	_	SUP reference:		
				clauses 7.7.1 3), 7.7.4, 7.7.5		
TSS reference	ISUP-SIP/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	<ul> <li>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message before a 200 OK response message has been received:</li> <li>the SUT shall hold the GRS message until a response has been received. A CANCEL is sent The GRS is not encapsulated;</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> </ul>					
SIP parameter values						
ISUP parameter values						
Comments	ISUP/BICC	SU	IT	SIP-I		
	IAM	<b>→</b>	<b>→</b>	INVITE(IAM)		
			+	100 Trying		
	GRS	<b>→</b>				
	GRA	+	→	CANCEL		
			+	200 OK INVITE(CON)		
			<b>→</b>	ACK		
			+	200 OK CANCEL		
			<b>→</b>	BYE(REL#31)		
			+	200 OK BYE(RLC)		

TP309011	SIP reference: RFC 3	261 [4]		I	SUP reference:		
	Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5						
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message						
	(GRS) or Circuit group blocking	ig message (C	CGB) with	the indi	cation hardware failure		
	oriented						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Ensure that the SUT after rece						
	sending a INVITE message w						
	message on receipt GRS mes						
	<ul> <li>the SUT shall send a BYE</li> </ul>	request The	GRS is no	ot encap	sulated.		
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP/BICC	;	SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		<b>←</b>	180 Ringing(ACM)		
	ANM ← 200 OK INVITE(ANM)						
	→ ACK						
	GRS	<b>→</b>		<b>→</b>	BYE(REL#31)		
	GRA	<b>+</b>		+	200 OK BYE(RLC)		

TP309012	SIP reference: RFC 3	261 [4]			ISUP reference:	
					], clauses 7.7.1, 7.7.4, 7.7.5	
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message					
	(GRS) or Circuit group blockir	ig mess	sage (CGI	B) with the ind	lication hardware failure	
	oriented					
SIP selection criteria						
ISUP selection						
criteria						
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established:  • the SUT shall send a CANCEL or BYE request The GRS is not encapsulated.					
SIP parameter values						
ISUP parameter						
values						
Comments	ISUP/BICC		SU		SIP-I	
	IAM	<b>→</b>		→	INVITE(IAM)	
				<b>←</b>	SIP_MESSAGE_VA	
	GRS	<b>→</b>				
	GRA	<b>←</b>				
	CASE A					
				→	CANCEL	
				<b>←</b>	200 OK CANCEL	
				<b>←</b>	487 Request terminated	
				<b>→</b>	ACK	
	CASE B					
				<b>→</b>	BYE(REL#31)	
				<b>←</b>	200 OK BYE(RLC)	
				<b>←</b>	487 Request terminated	
				→	ACK	

Table 18

	Values for test purpose TP309009; TP309012					
VA	SIP MESSAGE_VA					
VA_1	180 Ringing					
VA_2	183 Session Progress					

TP309013	SIP reference: RF	C 3261 [4]			ISUP reference: ], clauses 7.7.1, 7.7.4, 7.7.5	
TSS reference	ISUP-SIP/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	Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a GRS message were the Range Parameter value is bigger than "1":  the SUT shall send a BYE requests for each call association The GRS is not encapsulated.					
SIP parameter	BYE1 contains the CSeq of	of INVITE1				
values	BYE2 contains the CSeq of	of INVITE2				
ISUP parameter values						
Comments	ISUP/BICC		SUT	Ī I	SIP-I	
	IAM	<b>→</b>		→	INVITE1(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
				<b>→</b>	ACK	
	IAM	<b>→</b>		<b>→</b>	INVITE2(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
				→	ACK	
	GRS	<b>→</b>				
	GRA	+				
				<b>→</b>	BYE1(REL#31)	
				+	200 OK BYE(RLC)	
				<b>→</b>	BYE2(REL#31)	
				<del>(</del>	200 OK BYE(RLC)	

## 5.2.2.9.3 Receipt of Circuit group blocking message (CGB)

TP3090014	SIP reference: RFC 3261 [4] ISUP reference:						
		Q.1912.5 [1], clauses 7.7.1, 1), 7.7.4					
TSS reference	ISUP-SIP/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	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(hardware failure oriented)						
Comments	ISUP/BICC	SU	Т	SIP-I			
	IAM →						
	CGB →						
	CGBA <b>←</b>						

TP309015	SIP reference: RFC 3	261 [4]		SUP reference: [1], clauses 7.7.1, 7.7.4		
TSS reference	ISUP-SIP/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	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 The CGB is not encapsulated.					
SIP parameter values		•		•		
ISUP parameter values	CGB(hardware failure oriented	d)				
Comments	ISUP/BICC	SU	Т	SIP-I		
	IAM	<b>→</b>	<b>→</b>	INVITE(IAM)		
	CGB	<b>→</b>				
	CGBA	<b>←</b>				
			<del>-</del>	SIP_MESSAGE_VA		
	CASE A					
			→	CANCEL		
			←	200 OK CANCEL		
			<b>←</b>	487 Request terminated		
			<b>→</b>	ACK		
	CASE B	, , , , , , , , , , , , , , , , , , ,				
			<b>→</b>	BYE(REL#31)		
			<b>←</b>	200 OK BYE(RLC)		
			<b>←</b>	487 Request terminated		
			→	ACK		

Table 19

	Values for test purpose TP309015					
VA	SIP MESSAGE_VA					
VA_1	100 Trying					
VA_2	180 Ringing					
VA_3	183 Session Progress					

TP3090016	SIP reference: RFC 3261 [4]		_	SUP reference:				
<b>700</b> /				1], clauses 7.7.1 3), 7.7.4				
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message							
	oriented	(GRS) or Circuit group blocking message (CGB) with the indication hardware failure						
SIP selection criteria								
ISUP selection criteria								
Test purpose	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" <b>before</b> a 200 OK response message has been received:  on subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent The CGB is not encapsulated.							
SIP parameter values								
ISUP parameter values	CGB(hardware failure oriented)							
Comments	ISUP/BICC	SUT		SIP-I				
	IAM →		<b>→</b>	INVITE(IAM)				
			<b>←</b>	100 Trying				
	CGB →							
	CGBA <b>←</b>		<b>→</b>	CANCEL				
	← 200 OK INVITE(CO							
			<b>→</b>	ACK				
			+	200 OK CANCEL				
			<b>→</b>	BYE(REL#31)				
			<b>←</b>	200 OK BYE(RLC)				

TP309017	SIP reference: RFC 32	61 [4]	_	SUP reference:			
		Q.1912.5 [1], clauses 7.7.1, 7.7.4					
TSS reference	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message						
	(GRS) or Circuit group blocking	message (CG	B) with the indi	cation hardware failure			
	oriented						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" after a 200 OK response message has been received:  • the SUT shall send a BYE request The CGB is not encapsulated.						
SIP parameter values		·					
ISUP parameter values	CGB(hardware failure oriented)						
Comments	ISUP/BICC	SU	Т	SIP-I			
	IAM	<b>→</b>	<b>→</b>	INVITE(IAM)			
	ACM	<b>←</b>	+	180 Ringing(ACM)			
	ANM ← 200 OK INVÎTE(ANM)						
	→ ACK						
	002	<b>→</b>	<b>→</b>	BYE(REL#31)			
	CGBA	<b>←</b>	+	200 OK BYE(RLC)			

TP309018	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4			
TSS reference	ISUP-SIP/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	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" <b>after</b> an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established:  • the SUT shall send a CANCEL or BYE request The CGB is not encapsulated.						
SIP parameter values					·		
ISUP parameter values	CGB(hardware failure oriented)						
Comments	ISUP/BICC		SU	Т	SIP-I		
	IAM	→		→	INVITE(IAM)		
				<b>+</b>	SIP_MESSAGE_VA		
	CGB	<b>→</b>					
	CGBA	<b>←</b>					
	CASE A						
				<b>→</b>	CANCEL		
				+	200 OK CANCEL		
				+	487 Request terminated		
				→	ACK		
	CASE B						
		1		<b>→</b>	BYE(REL#31)		
		1		<del>-</del>	200 OK BYE(RLC)		
		1		<del>-</del>	487 Request terminated		
				→	ACK		

Table 20

Values for test purpose TP309014; TP309018					
VA	SIP MESSAGE_VA				
VA_1	180 Ringing				
VA_2	183 Session Progress				

TP309019	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 7.7.1, 7.7.4, 7.7.5				
TSS reference	ISUP-SIP/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	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" were the Range and Status Parameter value is bigger than "1":  the SUT shall send a BYE requests for each call association The CGB is not encapsulated.							
SIP parameter values	BYE1 contains the CSeq of INVITE1 BYE2 contains the CSeq of INVITE2							
ISUP parameter values	CGB(hardware failure oriented)							
Comments	ISUP/BICC		SUT		SIP-I			
	IAM	→		→	INVITE1(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	IAM	<b>→</b>		<b>→</b>	INVITE2(IAM)			
	ACM	<del>(</del>		+	180 Ringing(ACM)			
	ANM	←		<b>←</b>	200 OK INVITE(ANM)			
				→	ACK			
	CGB	→						
	CGBA	+						
				→	BYE1(REL#31)			
				+	200 OK BYE(RLC)			
				<b>→</b>	BYE2(REL#31)			
				+	200 OK BYE(RLC)			

## 5.2.2.10 Receipt of Confusion message

TP310001	SIP refer	ence: RFC 326	1 [4]	ISUP reference: Q.1912.5 [1], clauses A.1.1.3						
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Confusion message									
SIP selection criteria										
ISUP selection criteria										
Test purpose	Ensure that the SUT after receiving the IAM with the complete called party number and contains an unknown parameter, sending a INVITE message with the complete called party number and encapsulated IAM as received  Ensure that when the succeeding node discards an unknown parameter and send back a Confusion message if indicated in the parameter compatibility information and the sending of a Confusion message is requested, the CFN message encapsulated in a 183 Session Progress is sent.  Ensure ISUP message is transported through the SIP network encapsulated in the 183 Session Progress.									
SIP parameter values	183 Session Pro	ogress with enca	psulated CFI	N						
ISUP parameter values	CFN									
	ISUP				SIP-I					
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM with unknown parameter)					
	CFN	+		<b>←</b>	183 Session Progress(CFN)					
	ACM	+		4	180 Ringing(ACM)					
	ANM	+		← 200 OK INVITE(ANM)						
				→ ACK						
			nmunication							
	REL	→		<b>→</b>	BYE(REL)					
	RLC	+		<b>←</b>	200 OK BYE(RLC)					

## 5.2.2.11 Receipt of "Suspend" or "Resume" message

TP311001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause A.1.1.3				
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of Suspend message							
SIP selection criteria								
ISUP selection criteria								
Test purpose	<ul> <li>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, on receipt of a Suspend initiated by the network:</li> <li>ensure that the ISUP message is transported through the SIP network encapsulated in the INFO message;</li> <li>ensure that the called subscriber can successfully clear back and reanswer the call.</li> </ul>							
SIP parameter				,				
values								
ISUP parameter								
values								
Comments	ISUP/BICC		SUT	-	SIP-I			
	IAM	→		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
				→	ACK			
	Conversation							
	SUS	+		+	INFO(SUS)			
				<b>→</b>	200 OK INFO			
	RES	+		+	INFO(RES)			
				→	200 OK INFO			
		(	Conversa	ation				
	REL	<b>→</b>		→	BYE(REL)			
	RLC	+		<del>(</del>	200 OK BYE(RLC)			

## 5.2.2.12 Receipt of a Blocking message

TP312001	SIP reference: RFC 3261	[4]	ISUP reference: Q.1912.5 [1], clause A.1.1.3.1					
TSS reference	ISUP-SIP/ ISUP Messages for sp	ecial consid	eration / Rece	eipt of a Blocking message				
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the blocking/unblocki	ng procedure	e can be corre	ectly initiated. Ensure the BLO				
	messages is not encapsulated wi	thin SIP mes	sages					
SIP parameter								
values								
ISUP parameter								
values								
Comments	ISUP/BICC	SU	JT	SIP-I				
	BLO -	<b>&gt;</b>						
	BLA	<b>-</b>						
	UBL -	<b>&gt;</b>						
	UBA 🗲							

TP312002	SIP reference: RFC 326		ISUP reference: Q.1912.5 [1], clause A.1.1.3.1				
TSS reference	ISUP-SIP/ ISUP Messages for s	pecial conside	eration / Rece	eipt of a Blocking message			
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Ensure that the blocking from both ends; removal of blocking from one end can be correctly initiated. Ensure the BLO messages is not encapsulated within SIP messages.						
SIP parameter values							
ISUP parameter values							
Comments	ISUP/BICC	SU	JT	SIP-I			
	BLO	<b>→</b>					
	BLA	<b>←</b>					
	BLO	+					
	BLA →						
	UBL	<b>→</b>					
	UBA	+					

TP312003	SIP reference: RFC 3261 [4]		ISUP reference:				
		Q.1912.5 [1], clause A.1.1.3.1					
TSS reference	ISUP-SIP/ ISUP Messages for	special consi	deration / Rece	eipt of a Blocking message			
SIP selection							
criteria							
ISUP selection criteria							
Test purpose	CGB and CGU sent						
	Ensure that the SUT is able to	respond on a	Circuit group b	locking message with a CGBA			
	and on a Circuit group unblocki	ing message	both maintena	ance oriented) with a CGUA.			
	Ensure the CGB / CGU message	ges are not e	ncapsulated wi	thin SIP messages.			
SIP parameter							
values							
ISUP parameter							
values							
Comments	ISUP	5	UT	SIP-I			
	CGB	CGB →					
	CGBA ←						
	CGU	<b>→</b>					
	CGUA	+					

TP312004	SIP reference: RFC 3261 [4]		ISUP reference: 2.5 [1], clause A.1.1.3.1						
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a Blocking message								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT on receipt of a CGB, messages, discards the ISUP information		encapsulated within SIP						
SIP parameter values									
ISUP parameter values									
Comments	ISUP	SUT	SIP-I						
		+	INFO(CBG)						

TP312005	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause A.1.1.3.1 Q.784 [i.11], clause 1.3.2.4		
TSS reference	ISUP-SIP/ ISUP Messa	ges for special	conside	eration /	Receip	ot of a Blocking message
SIP selection					•	
criteria						
ISUP selection						
criteria						
Test purpose	Ensure that a received	IAM will unblock	k a rem	otely blo	cked c	ircuit.
SIP parameter				-		
values						
ISUP parameter						
values						
Comments	ISUP		SU	JT		SIP-I
	BLO	<b>→</b>				
	BLA	+				
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)
	ACM	+			+	180 Ringing(ACM)
	ANM	+			+	200 OK INVITE(ANM)
					<b>→</b>	ACK
	REL	<b>→</b>			<b>→</b>	BYE(REL)
	RLC	+			+	200 OK BYE(RLC)

## 5.2.2.13 Receipt of a user part test message

TP313001	SIP reference: RFC	3261 [4]	_	SUP reference: 5 [1], clause A.1.1.3.1					
			Q.784	[i.11], clause 1.3.2.4					
TSS reference	ISUP-SIP/ ISUP Messages f	or special consid	eration / Receip	t of a user part test message					
SIP selection criteria									
ISUP selection criteria	PICS 4/22	PICS 4/22							
Test purpose	Ensure that on receipt of a user part test message the SUT will respond by sending a user part available message.  Ensure that the user part test message is not encapsulated within SIP messages.								
SIP parameter values	and the same of th								
ISUP parameter values									
Comments	ISUP	Sl	JT	SIP-I					
	UPT	<b>→</b>							
	UPA	+							

TP313002	SIP reference: RF0	C 3261 [4]	Q	ISUP reference: .1912.5 [1], clause A.1.1.3.1					
TSS reference	ISUP-SIP/ ISUP Messages for special consideration / Receipt of a user part test message								
SIP selection criteria									
ISUP selection	PICS 4/22	PICS 4/22							
criteria									
Test purpose	Ensure that the SUT is able	to send a us	er part test me	ssage.					
SIP parameter									
values									
ISUP parameter									
values									
Comments	ISUP	ISUP SUT SIP-I							
	UPT	+							
	UPA	<b>→</b>							

TP313003	SIP reference: RFC 326	erence: RFC 3261 [4] ISUP reference:					
			Q.1912.5	[1], clause A.1.1.3.1			
TSS reference	ISUP-SIP/ ISUP Messages for s	pecial conside	eration / Receipt of	of a user part test message			
SIP selection criteria							
ISUP selection criteria	PICS 4/22						
Test purpose	T4 Waiting to receive a response to a user part test message.  Ensure that the SUT is able to restart the availability test procedure after expiry of timer T4.						
SIP parameter values							
ISUP parameter							
values							
Comments	ISUP	SL	JT	SIP-I			
	UPT ←						
	T4 expiry						
	UPT •	L					
	UPA -	<b>→</b>					

## 5.2.2.14 Segmentation

TP314001	SIP reference: RFC	SIP reference: RFC 3261 [4]			ISUP reference:			
	Q.1912.5 [1], clause A.1.1.3.1							
TSS reference	ISUP-SIP/ ISUP Messages	for spe	cial conside	eration / Re	ceip	t of a user part test message		
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that a call can be su direction.	Ensure that a call can be successfully completed if segmentation applies in forward direction.						
SIP parameter values	INVITE - encapsulated IAM: information will be sent" No action takes place on the							
ISUP parameter values	IAM: optional forward call in message SGM: optional parameters			information	will	be sent in a segmentation		
Comments	ISUP		SU	Т		SIP-I		
	IAM	<b>→</b>						
	SGM	<b>→</b>			<del>→</del>	INVITE(IAM)		
	ACM	+			<del>(</del>	180 Ringing(ACM)		
	ANM	+			<del>(</del>	200 OK INVITE(ANM)		
	→ ACK							
			Convers	ation				
	REL	<b>→</b>			<b>→</b>	BYE(REL)		
	RLC	+			<del>(</del>	200 OK BYE		

# 5.3 Test purposes for the Supplementary Services

## 5.3.1 Calling Line Identification Presentation (CLIP)

TP401001	SIP reference: RFC	3261	[4]		Į;	SUP reference:			
					Q.191	2.5 [1], clause 7.1.3			
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
ISUP selection									
criteria									
Test purpose	Calling Party number netwo	Calling Party number network provided, transferred in O-MGCF							
		Ensure that the SUT can successfully transmit a call having a <b>calling party number</b> with the screening indicator set to "network provided" and the presentation restricted indicator set to "presentation allowed".							
SIP parameter									
values									
ISUP parameter	IAM;								
values	Calling party number para	meter							
	Address signals = PIXIT1								
	Numbering plan indicator =								
	Nature of address indicator	= '0000'	0011'B						
	Screening indicator = '11'B								
_	presentation restricted indic	ator = p			'00'B				
Comments	ISUP		SU			SIP-I			
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ACM	+			+	180 Ringing(ACM)			
	ANM	+			+	200 OK INVITE(ANM)			
			Convers	ation					
	REL	<b>→</b>			<b>→</b>	BYE(REL)			
	RLC	<b>←</b>			+	200 OK BYE(RLC)			

TP401002	SIP reference: RFC	3261	[4]	_	SUP reference: 2.5 [1], clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Calling Party number networkprovided, Calling Subaddress transferred in O-MGCF								
	Ensure that the SUT can su the screening indicator set t containing the <b>calling sub-</b>	o "netw	ork provided" an		calling party number with ess transport parameter				
SIP parameter									
values									
ISUP parameter									
values	IAM;								
	Calling party number para	meter							
	Address signals = PIXIT1								
	Numbering plan indicator =								
	Nature of address indicator	= '0000	0011'B						
	Screening indicator = '11'B			LIGOIE					
	presentation restricted indic								
0	Access transport parameter	includi		ss morma					
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	<del>(</del>		<del>-</del>	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
			Conversation						
	REL	<b>→</b>		→	BYE(REL)				
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)				

TP401003	SIP reference: RFC	3261	[4]			SUP reference:
<del></del>	10110 010 10110 00 10110				J.191	2.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Calling Party Number user p	nrovide	d transferred	in O-MG	CF	
rest purpose	Calling Farty Number user p	Jiovide	a transierrea	III O-IVIC	<i>J</i> 01	
	Ensure that the SUT can su	ccessfi	ully transmit a	a call hav	/ina th	ne calling party number with
	the screening indicator set t					
	restricted indicator set to "pi					
SIP parameter						
values						
ISUP parameter	IAM;					
values	Calling party number para	meter				
	Address signals = PIXIT1					
	Numbering plan indicator =	'001'B				
	Nature of address indicator	= '0000'	0011'B			
	Screening indicator = '01'B					
	presentation restricted indic	cator =		allowed,	'00'E	
Comments	ISUP		SUT			SIP-I
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)
	ACM	+			<del>-</del>	180 Ringing(ACM)
	ANM	+			<b>←</b>	200 OK INVITE(ANM)
			Conversat	ion		
	REL	<b>→</b>			<b>→</b>	BYE(REL)
	RLC	<b>←</b>			<b>←</b>	200 OK BYE(RLC)

TP401004	SIP reference: RFC 3	3261	[4]	-	SUP reference:
				Q.191	2.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Calling Party Number user pr	ovide	d and calling su	baddress t	ransferred in O-MGCF
Ì	Ensure that the SUT can succ				
	the screening indicator set to				sed" and an <b>access</b>
	transport parameter containi	ing the	e calling sub-a	ddress.	
SIP parameter					
values					
ISUP parameter	IAM;				
values	Calling party number paran	neter			
	Address signals = PIXIT1				
	Numbering plan indicator = '0				
	Nature of address indicator =	'0000	011'B		
	Screening indicator = '01'B				
	Presentation restricted indica				
	Access transport parameter in	ncludi		ess informa	
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		→	INVITE(IAM)
	ACM	<b>←</b>		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversatio	n	
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP401005	SIP reference: RFC	3261	[4]			SUP reference:					
					Q.191	2.5 [1], clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP										
SIP selection criteria											
ISUP selection											
criteria											
Test purpose	Calling Party Number network provided and additional calling party number user provided not verified transferred in O-MGCF.										
	Ensure that the SUT can sunumber with the screening containing the additional cal provided, not verified and thallowed.	indicato ling pa	or set to "net rty number v	work pro vith the s	vided' creen	and a <b>generic number</b> ing indicator set to "user					
SIP parameter											
values											
ISUP parameter values	IAM;										
	Calling party number para	meter									
	Address signals = PIXIT1										
	Numbering plan indicator =	'001'B									
	Nature of address indicator		0011'B								
	Screening indicator = '11'B										
	Presentation restricted indic	ator =	presentation	allowed,	'00'B						
	Generic number paramete	er									
	Address signals = PIXIT2										
	Numbering plan indicator =										
	Nature of address indicator	= '0000	0011'B								
	Screening indicator = '00'B	_4		-11	IOOID						
0	Presentation restricted indic	ator =			.00.B						
Comments	ISUP	_	SUT	1		SIP-I					
	IAM ACM	<b>→</b>			<u>→</u>	INVITE(IAM)					
	ACM	4			<del>-</del>	180 Ringing(ACM) 200 OK INVITE(ANM)					
	AINIVI		Conversa	tion		ZUU UK IINVITE(AINIVI)					
	DEL	_	Conversa	itiOff	_	DVE(DEL)					
	REL	<b>→</b>		+	<u>→</u>	BYE(REL)					
	RLC	7			_	200 OK BYE(RLC)					

TP401006	SIP reference: RFC	3261	[4]		ISUP reference: 12.5 [1], clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP				[.],					
SIP selection	1001 011 1001 7007 0211									
criteria										
ISUP selection										
criteria										
Test purpose		Calling Party Number network provided, additional calling party number user provided not verified and calling subaddress transferred in O-MGCF.								
	with the screening indicator additional calling party numb	Ensure that the SUT can successfully transmit a call having a default calling party number with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified" and an access transport parameter containing the calling sub-address.								
SIP parameter values					<u> </u>					
ISUP parameter values	IAM;	IAM;								
	Calling party number para	meter								
	Address signals = PIXIT1 Numbering plan indicator = ' Nature of address indicator = Screening indicator = '11'B		0011'B							
	Generic number parameter	r								
	Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Access transport parameter including the subaddress information									
Comments	ISUP		SU		SIP-I					
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ACM	+		<b>←</b>	180 Ringing(ACM)					
	ANM	+		+	200 OK INVITE(ANM)					
			Convers	ation						
	REL	<b>→</b>		<b>→</b>	BYE(REL)					
	RLC	+		+	200 OK BYE(RLC)					

TP401007	SIP reference: RF	C 3261 [4	]		ISUP reference:				
				Q.1912.5 [1], clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
ISUP selection criteria	PICS 6/8								
Test purpose	Calling party number disca	arded to du	ue bilateral a	agreement in	the I-MGCF.				
	Ensure that the calling par address presentation restr				oilateral agreements, if the on allowed" (see note).				
SIP parameter values									
ISUP parameter	IAM;								
values	No calling party number	paramete	r						
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
		Conversation							
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				
	teral agreement prohibits the address presentation restrict			• • •	-				

TP401008	SIP reference: RFC	3261	[4]		ISUP reference:
				Q.19	112.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria	PICS 6/7				
Test purpose		alling p	arty numbe	r in the <b>gener</b> i	agreements in the I-MGCF ic number is discarded in case d indicator is set to
SIP parameter values					
ISUP parameter	IAM;				
values	No calling party number p	arame	ter		
Comments	SIP-I		SU	Γ	ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Convers	ation	
	BYE(REL)	<b>→</b>		→	REL
	200 OK BYE(RLC)	+		+	RLC
	eral agreement prohibits the t ss presentation restricted ind				mber in any case. The test with ted" is a CLIR test.

TP401009	SIP reference: RF	C 3261	[4]		-	SUP reference:
				Q	.191	2.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP					
SIP selection criteria						
ISUP selection criteria	PICS 6/6					
Test purpose	Calling party number is orn available in the I-MGCF Ensure that the calling pai indicator is set to "address	rty num	n <b>ber</b> is omitt			indicator is set to address not ss presentation restricted
SIP parameter						
values						
ISUP parameter						
values						
Comments	SIP-I		SU	Γ		ISUP
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM
	180 Ringing(ACM)	+		•	←	ACM
	200 OK INVITE(ANM)	+		1	<del>(</del>	ANM
			Convers	ation		
	BYE(REL)	→		•	<del>&gt;</del>	REL
	200 OK BYE(RLC)	+		•	<del>(</del>	RLC

TP401010	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.1.3		
TSS reference	ISUP-SIP-ISUP/SS/CLIP					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Calling party number is se	nt as red	ceived			
SIP parameter values ISUP parameter values	Ensure that the calling par number in the encapsulate		er in the sent IAN	is gener	ated from the calling party	
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM	
	180 Ringing(ACM)	+		+	ACM	
	200 OK INVITE(ANM)	+		+	ANM	
			Conversation	-		
	BYE(REL)	<b>→</b>		<b>→</b>	REL	
	200 OK BYE(RLC)	+		+	RLC	

TP401011	SIP reference: RF	C 3261	[4]		SUP reference:
				Q.19 <sup>-</sup>	12.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Additional calling party nu	mber is s	sent as receiv	ed	
	Ensure that the additional				M is generated from the
	additional calling party nur	mber in t	he encapsulat	ed IAM.	
SIP parameter					
values					
ISUP parameter					
values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversat	ion	
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP401012	SIP reference: RF	C 3261	[4]		ISUP reference:
				Q.19 <sup>-</sup>	12.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Additional calling party nul	mber is d	omitted in the	I-MGCF	
SID parameter	Ensure that if the <b>calling</b> pumber in a <b>generic num</b>	<b>ber</b> will b	oe omitted.		
SIP parameter	9.	umber in	iciuaea in the	encapsulate	d IAM, additional calling party
values	number included.				
ISUP parameter	IAM;				
values	No calling party number	parame	ter		
	No generic number para	meter			
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Conversat	ion	
	BYE(REL)	<b>→</b>		→	REL
	200 OK BYE(RLC)	+		+	RLC

TP401013	SIP reference: RFC	3261	[4]			SUP reference: 31 [i.2], clause 3.5				
TSS reference	ISUP-SIP-ISUP/SS/CLIP									
SIP selection										
criteria										
ISUP selection	PICS 1/7									
criteria										
Test purpose	Convert the Calling party number into the international format in the I-MGCF									
	Ensure that the SUT can co									
	setting the nature of address									
CID novemeter	address presentation restric	tea ina	icator and ti	ne screer	ning in	dicator transparently.				
SIP parameter values										
	IAM:									
ISUP parameter values	Calling party number para	motor								
values	Address signals = PIXIT1	meter								
	Numbering plan indicator =	'001'B								
	Nature of address indicator		)100'B							
	Screening indicator = '11'B	_ 0000	7.002							
	Presentation restricted indic	ator =p	resentation	allowed,	'00'B					
Comments	SIP-I	·	SUT			ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	180 Ringing(ACM)	+			+	ACM				
	200 OK INVITE(ANM)	<b>+</b>			+	ANM				
			Convers	ation						
	BYE(REL)	<b>→</b>			<b>→</b>	REL				
	200 OK BYE(RLC)	4			<b>+</b>	RLC				

TP401014	SIP reference: RFC	3261	[4]			SUP reference: 3.5/Q.731 [i.2]			
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
ISUP selection	PICS 1/7								
criteria									
Test purpose	Converting the additional calling party number to international format in the I-MGCF								
	Ensure that the SUT can conumber into an international setting the nature of address address presentation restricts.	al numb s indica	er, if the nu ator to "inter	mbering pl national nu	lan ir umbe	ndicator is "ISDN Telephony", er" and can pass on the			
SIP parameter					· · · · ·				
values									
ISUP parameter	IAM								
values	Address signals = PIXIT1 Numbering plan indicator = Nature of address indicator Screening indicator = '11'B Presentation restricted indic Generic number paramete Address signals = PIXIT2 Numbering plan indicator = Nature of address indicator Screening indicator = '00'B Presentation restricted indic	Calling party number parameter  Address signals = PIXIT1  Numbering plan indicator = '001'B  Nature of address indicator = '0000100'B  Screening indicator = '11'B  Presentation restricted indicator =presentation allowed, '00'B  Generic number parameter  Address signals = PIXIT2  Numbering plan indicator = '001'B  Nature of address indicator = '0000100'B							
Comments	SIP-I		SUT			ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	+			<del>(</del>	ACM			
	200 OK INVITE(ANM)	+			<del>(</del>	ANM			
			Conversa						
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			<del>(</del>	RLC			

TP401015	SIP reference: RFC 3261 [4]				ISUP reference: 3.5/Q.731 [i.2]			
TSS reference	ISUP-SIP-ISUP/SS/CLIP							
SIP selection criteria								
ISUP selection criteria	PICS 1/7 AND NOT PICS	1/9						
Test purpose	Ensure that the calling par	Discarding an incomplete calling party number in the I-MGCF  Ensure that the calling party number is discarded, if it is received with the calling party number incomplete indicator set to "incomplete" (see note).						
SIP parameter values								
ISUP parameter values	IAM: No calling party number	parame	ter					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	, ,	Conversation						
	BYE(REL)	→		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			
NOTE: This test	case is only applicable with	an ITU i	mplementation.	•				

TP401016	SIP reference: RFC 3261 [4] ISUP reference: 3.5/Q.731 [i.2]										
TSS reference	ISUP-SIP-ISUP/SS/CLIP										
SIP selection											
criteria											
ISUP selection	PICS 1/8										
criteria											
Test purpose	Converting the calling party	numbe	er to national	format,	if nece	essary in the O-MGCF					
	Ensure that the country code										
	removed if it is the network's										
	, ,	numbe	r". The addre	ess pres	entatio	on restricted indicator shall be					
OID	transferred transparently.										
SIP parameter	INVITE: encapsulated IAM										
values	Calling party number		ımeter								
	Address signals = PI		1004 ID								
	Numbering plan indic Nature of address ind			D							
	Screening indicator =		= 00000111	Ь							
	Presentation restricte		rator – prese	ntation a	allowe	d '00'B					
ISUP parameter	IAM	o maic	bator = prose	mation	anowe	<u>a, 00 B</u>					
values	Calling party number param	eter									
	Address signals = PIXIT1										
	Numbering plan indicator =	001'B									
	Nature of address indicator		0100'B								
	Screening indicator = '11'B										
	Presentation restricted indic	ator =	presentation	allowed	, '00'B						
Comments	SIP-I		SUT			ISUP					
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)					
	ACM	<b>←</b>			+	180 Ringing(ACM)					
	ANM	<del>(</del>			+	200 OK INVITE(ANM)					
		-	Conversa	tion							
	REL	<b>→</b>			<b>→</b>	BYE(REL)					
	RLC	<b>←</b>			+	200 OK BYE(RLC)					

TP401017	SIP reference: RFC	3261	[4]		-	SUP refere			
TSS reference:	3.5/Q.731 [i.2] ISUP-SIP-ISUP/SS/CLIP								
SIP selection	100. 01. 100.700/0211								
criteria									
ISUP selection criteria	PICS 1/8								
Test purpose	Converting the additional calling party number to national format, if necessary in the O-MGCF								
	"additional calling party num removed if it is the network's	Ensure that the country code in the address signals of the <b>generic number</b> coded as an "additional calling party number", if the numbering plan indicator is "ISDN Telephony" is removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number". The address presentation restricted indicator shall be							
SIP parameter	INVITE: encapsulated IAM								
values	Generic number par	amete	r						
	Address signals = PI								
	Numbering plan indic								
	Nature of address inc		= '0000011	В					
	Screening indicator =								
	Presentation restricte	d indic	cator = prese	entation a	llowe	d, '00'B			
ISUP parameter	IAM;								
values	Calling party number para	meter							
	Address signals = PIXIT1 Numbering plan indicator = '	001'D							
	Nature of address indicator =		0011'B						
	Screening indicator = '11'B	= 0000	МПБ						
	Presentation restricted indica	ator –	nresentation	allowed	'00'B				
	Generic number paramete		presentation	i anowca,	000	'			
	Address signals = PIXIT2								
	Numbering plan indicator = '	001'B							
	Nature of address indicator =		0011'B						
	Screening indicator = '00'B								
	Presentation restricted indica	ator =	presentation	allowed,	'00'B				
Comments	SIP-I		SUT	•		ISUP			
	IAM	<b>→</b>			<b>→</b>	INVITE(IAI			
	ACM	+			<del>-</del>	180 Ringin			
	ANM	+			<del>-</del>	200 OK IN	VITE(ANM)		
			Convers	ation					
	REL	<u>→</u>			<b>→</b>	BYE(REL)			
	RLC	+			<del>-</del>	200 OK BY	'E(RLC)		

TP401018	SIP reference: RFC 3261 [4]				ISUP reference: 3.5/Q.731 [i.2]			
TSS reference	ISUP-SIP-ISUP/SS/CLIP				-			
SIP selection criteria								
ISUP selection criteria	PICS 1/7							
Test purpose		Adding a prefix to an international calling party number in the I-MGCF  Ensure that a prefix is added to the <b>calling party number</b> and the nature of address indicator is set to "unknown" (see pate)						
SIP parameter values		`	,					
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation	•				
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			
NOTE: The codir	ng "unknown" is a national o	ption (@	.).					

TP401019	SIP reference: RF0	3261	[4]	ı	SUP reference: 3.5/Q.731 [i.2]				
TSS reference	ISUP-SIP-ISUP/SS/CLIP				5.6/Q.751 [I.2]				
SIP selection	1001 -011 -1001 /00/0211								
criteria									
ISUP selection									
criteria									
Test purpose	MGCF								
	Ensure that the screening in presentation restricted indicavailable" (see note).								
SIP parameter									
values									
ISUP parameter	IAM;								
values	Calling party number para	ameter							
	Address signals = PIXIT1								
	Numbering plan indicator =	' *'B							
	Nature of address indicator	= '*'B							
	Screening indicator = '11'B								
	Presentation restricted indicate	cator =a	ddress not av	ailable, '10'E	3				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		→	IAM				
	180 Ringing(ACM)	+		<del>-</del>	ACM				
	200 OK INVITE(ANM)	+		<del>-</del>	ANM				
			Conversati	on					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		<del>-</del>	RLC				
NOTE: The codir	ng "address not available" is	a natior	al option (@).	•					

TP401020	SIP reference: RFC	3261	[4]	_	SUP reference: 12.5 [1], clause 7.1.3
TSS reference	ISUP-SIP-ISUP/SS/CLIP				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	O-MGCF: Calling party nun	nber an	d Additional callir	ng party n	number not received
SIP parameter values ISUP parameter values	parameter and the Generic Sends an INVITE message	Numbe withou nous@a ntity, Fr	er are not applical t the "P-Asserted anonymous.invali om Header: anor	ole. -Identity h d". No Pr nymous@	rivacy header field included. eanonymous.inv
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	<b>←</b>		+	200 OK INVITE(ANM)
			Conversation	•	, , ,
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP401021	SIP reference: RFC 3261 [4] ISUP reference: Q.1912.5 [1], clause 7.1.3									
TSS reference:	ISUP-SIP-ISUP/SS/CLIP									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	O-MGCF: Setting of From h	eader								
SIP parameter values	parameter is <b>not applicable</b> presentation restriction para Address Indicator is set to N Sends an INVITE message field" where the user portion number and the country country "+"CC+NDC+SN and no "Pi	e and the ameter loAS_\ without of the le is serivacy hatity, no ber	ne Generic Num is set to "present /ALUE. t the "P-Asserted addr-spec is set t to the country theader field". Privacy header	tation allow d-Identity h to value o where the	neader field", a "From header of the additional calling party MGCF is located in the format ader contains the value of the					
values	ICUD		CUT		CID I					
Comments	ISUP		SUT		SIP-I					
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ACM	+		<del>-</del>	180 Ringing(ACM)					
	ANM ← 200 OK INVITE(ANM)									
			Conversation							
	REL	<b>→</b>		→	BYE(REL)					
	RLC ← 200 OK BYE(RLC)									

TP401022	SIP reference: RFC	3261	[4]	=	SUP reference:				
<i>(</i>	LOUID OID IOUD/OO/OUD			Q.191	2.5 [1], clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	O-MGCF: Setting of P-Asse Ensure that when the SUT h			nessage, the	e Calling Party Number is				
	applicable whereby the Natset to presentation allowed Sends an INVITE message	ture of and the	Address Indica	tor is set to	NoAS_VALUE the APRI is				
	the "P-Asserted-Identity header field" where the user portion of the addr-spec is set to value of the calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN;								
		e of the	e additional call IGCF is located	ing party nu I in the form	ere the user portion of the umber and the country code is eat "+"CC+NDC+SN;				
SIP parameter	INVITE: P-Asserted-Identity				mber, Privacy=id, From				
values	header derived from the add	ditional	calling party nu	ımber	•				
ISUP parameter	IAM; Calling party number is	s prese	nt and no Addit	tional calling	g party number is present				
values					le				
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		<b>+</b>	180 Ringing(ACM)				
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)				
			Conversation	n					
	REL	1		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP401023	SIP reference: RFC	3261	[4]			SUP reference:			
T00 (	10115 015 10115 (00 (01 15			(	≀.191	2.5 [1], clause 7.1.3			
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
ISUP selection criteria									
Test purpose	O-MGCF: Setting of P-Asserted header header and From header								
	Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation allowed and the Generic Number is applicable  Sends an INVITE message with:  • the "P-Asserted-Identity header field", " where the user portion of the addr-spec is set to value of the calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN;  • "From header field" " where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN;  • and without "Privacy Header field" or "id" is not included.								
SIP parameter	INVITE: P-Asserted-Identity	derive	d from the o	alling part	y nur	nber, no Privacy header,			
values	From header derived from the								
ISUP parameter values	IAM; Calling party number a	nd Add	ditional callin	ng party nu	ımbe	r are present			
Comments	ISUP		SUT	Γ		SIP-I			
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ACM	+			<del>(</del>	180 Ringing(ACM)			
	ANM	<del>-</del>			+	200 OK INVITE(ANM)			
			Convers	ation					
	REL	<b>→</b>			<b>→</b>	BYE(REL)			
	RLC	+			<del>(</del>	200 OK BYE(RLC)			

	Values for test purpose TP401021, TP401022, TP401023							
NoAS_VALUE	ISUP parameter values	SIP parameter values:						
VA_01	IAM	INVITE						
	NoAS_VALUE: "national (significant) number"(NDC+SN)	FHf_Addr_SPEC_ID: CC (of the country where the IWU 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)	FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used						

TP401024	SIP reference: RFC	3261	[4]		-	SUP reference:  2.5 [1], clause 7.1.3			
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
ISUP selection criteria	PICS 1/7								
Test purpose	Calling party derived from the P-Asserted-Identity international number								
	Privacy value "id" received.	ncapsu	lated IAM is	s not iden	tical to	o the P-Asserted-Identity, no			
	Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation allowed.								
SIP parameter values	INVITE: P-Asserted identity "id" is not present	user p	ortion is in t	he format	t "+"C(	C+NDC+SN, Privacy value			
ISUP parameter	IAM message with the Calli	ng par	ty number	paramet	er coc	led			
values	Address signals = nu Screening indicator = Number Incomplete I Numbering plan indic Address Presentation NoAS: "international	= netwo Indicato cator = n Restr	ork provided or = PIXIT ISDN numb icted Indica	l pering pla	n	·			
Comments	SIP-I		SU	Γ		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
			Convers	ation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP401025	SIP reference: RFC	3261	[4]	C	_	SUP reference: 12.5 [1], clause 7.1.3			
TSS reference	ISUP-SIP-ISUP/SS/CLIP								
SIP selection criteria									
ISUP selection criteria	NOT PICS 1/7								
Test purpose	Calling party derived from the P-Asserted-Identity national (significant) number								
	Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, no Privacy value "id" received.  Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation allowed.								
SIP parameter	INVITE: P-Asserted identity	user p	ortion is in tl	he format	"+"C	C+NDC+SN, Privacy value			
values	"id" is not present								
ISUP parameter	IAM message with the Callin								
values	Address signals = nu				serte	ed-Identity			
	Screening indicator =								
	Number Incomplete I								
	Numbering plan indic								
	Address Presentation			tor = Pres	entat	tion allowed			
	NoAS: "national (sign	nificant							
Comments	SIP-I		SUT			ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	<b>←</b>			<b>←</b>	ACM			
	200 OK INVITE(ANM)	<b>←</b>			<b>←</b>	ANM			
			Conversa	ation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			<b>←</b>	RLC			

TP401026	SIP reference: RFC	3261	[4]			SUP reference:					
T00 (	IOLID OID IOLID/OO/OLID				Q.191	2.5 [1], clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIP	UF-3 F- 3UF/33/CL F									
SIP selection											
criteria											
ISUP selection	PICS 1/7	ICS 1/7									
criteria											
Test purpose	Additional calling party number derived from the From header international number										
	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, no Privacy value "id" received.  Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation allowed.										
SIP parameter	INVITE: P-Asserted identity	user po	ortion is in t	he format	t "+"C(	C+NDC+SN, Privacy value					
values	"id" is not present										
ISUP parameter	IAM message with the Addi	tional	Calling par	ty numb	er par	ameter coded					
values	Address signals = nu					der					
	Screening indicator =			ot verified	"						
	Number Incomplete I	ndicato	or = PIXIT								
	Numbering plan indic										
	Address Presentation	n Restr	icted Indica	tor = Pres	sentat	ion allowed					
	NoAS: "international	numbe	r"								
Comments	SIP-I		SUT			ISUP					
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM					
	180 Ringing(ACM)	+			+	ACM					
	200 OK INVITE(ANM)	+			+	ANM					
	, , ,		Convers	ation							
	BYE(REL)	<b>→</b>			<b>→</b>	REL					
	200 OK BYE(RLC)	+			+	RLC					

TP401027	SIP reference: RFC	3261 [	[4]	0	-	SUP reference:  2.5 [1], clause 7.1.3				
TSS reference:	ISUP-SIP-ISUP/SS/CLIP			<u>u</u>	. 131	12.5 [1], clause 1.1.5				
SIP selection criteria										
ISUP selection criteria	NOT PICS 1/7									
Test purpose	Additional calling party number derived from the From header national (significant) number  Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the encapsulated IAM is not identical to the From header field, no Privacy value "id" received.  Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation allowed.									
SIP parameter	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value									
values	"id" is not present									
ISUP parameter	IAM message with the Addit									
values	Address signals = nun Screening indicator = Number Incomplete In	User	provided, no		hea	der				
	Numbering plan indica Address Presentation NoAS: "national (signi	ator = Restr	ISDN numbe icted Indicat		entat	tion allowed				
Comments	SIP-I		SUT			ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	180 Ringing(ACM)	+		•	<del>(</del>	ACM				
	200 OK INVITE(ANM)	+		•	<del>(</del>	ANM				
	, , ,	L. C.	Conversa	tion						
	BYE(REL)	<b>→</b>			<del>&gt;</del>	REL				
	200 OK BYE(RLC)	<b>←</b>			<del>(</del>	RLC				

# 5.3.2 Calling Line Identification Restriction (CLIR)

TP402001	SIP reference: RFC	3261	[4]	Q.73	ISUP reference: Q.1912.5 [1] 1 [i.2], clause 4.5.2.1.1				
TSS reference:	ISUP-SIP-ISUP/SS/CLIR								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Calling party number netwo	rk prov	ided presenta	ntion restrict	ted is passed.				
	Ensure that the SUT can successfully transmit a call having a <b>calling party number</b> with the screening indicator set to "network provided" and the address presentation restricted indicator set to "presentation restricted".								
SIP parameter									
values									
ISUP parameter	IAM;								
values	Calling party number para Screening indicator = '11'B Address presentation restric Generic number paramete Access transport paramet	ted pa r not p	rameter = '01 resent		ess information				
Comments	ISUP		SUT		SIP-I				
	IAM	<b>^</b>		→	INVITE(IAM)				
	ACM	+		←	180 Ringing(ACM)				
	ANM ← 200 OK INVITE(ANM)								
			Conversat						
	REL	<b>→</b>		→	BYE(REL)				
	RLC	+		←	200 OK BYE(RLC)				

TP402002	SIP reference: RFC	3261	[4]		Į,	SUP reference:						
						Q.1912.5 [1],						
					2.731	[i.2], clause 4.5.2.1.1						
TSS reference	ISUP-SIP-ISUP/SS/CLIR											
SIP selection												
criteria												
ISUP selection												
criteria												
Test purpose	Restricted calling party nu											
						alling party number with the						
						esentation restricted indicator						
		d" and	an ${\it access}$	transpo	rt para	ameter containing the <b>calling</b>						
	sub-address.	sub-address.										
SIP parameter												
values												
ISUP parameter	IAM;											
values	Calling party number parar	meter										
	Screening indicator = '11'B											
	Address presentation restrict			1'B								
	Generic number parameter											
	Access transport parameter	er inclu			ormati							
Comments	ISUP		SUT			SIP-I						
	IAM	<u>→</u>			<b>→</b>	INVITE(IAM)						
	ACM	<u>+</u>			+	180 Ringing(ACM)						
	ANM	+			<b>←</b>	200 OK INVITE(ANM)						
		Conversation										
	REL	<b>→</b>			<b>→</b>	BYE(REL)						
	RLC	+			+	200 OK BYE(RLC)						

TP402003	SIP reference: RFC	3261	[4]	_	SUP reference: Q.1912.5 [1], [i.2], clause 4.5.2.1.1				
TSS reference:	ISUP-SIP-ISUP/SS/CLIR		l l		[],				
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT can pa screening indicator set to "u	Restricted calling party number (user provided, verified and passed) Ensure that the SUT can pass transparently a call having the calling party number with the screening indicator set to "user provided, verified and passed" and the address presentation restricted indicator set to "presentation restricted".							
SIP parameter values									
ISUP parameter	IAM								
values	Calling party number para Address signals = PIXIT1 Numbering plan indicator = Nature of address indicator Screening indicator = '01'B Address presentation restrict	'001'B = '0000	rameter = '01'B						
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		<b>←</b>	200 OK INVITE(ANM)				
			Conversation						
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	<b>←</b>		<b>+</b>	200 OK BYE(RLC)				

TP402004	SIP reference: RFC	3261	[4]		IS	SUP reference: Q.1912.5 [1],					
				Q	.731	[i.2], clause 4.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/CLIR										
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	Restricted calling party nu	ımber	(user provi	ded, verif	fied a	nd passed) with calling					
	sub-address										
						alling party number with the					
	screening indicator set to "u										
	restricted indicator set to "pi containing the calling sub-a			eu anu ai	acc	ess transport parameter					
SIP parameter	containing the <b>calling sub-</b>	auures	3.								
values											
ISUP parameter	IAM										
values	Calling party number para	meter									
	Address signals = PIXIT1										
	Numbering plan indicator =	'001'B									
	Nature of address indicator		0011'B								
	Screening indicator = '01'B										
	Address presentation restric	ted pa	rameter = 'C	)1'B							
	Access transport paramet	er inclu	uding subad	ldress info	rmati	on					
Comments	ISUP		SUT	「		SIP-I					
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)					
	ACM	+			<del>-</del>	180 Ringing(ACM)					
	ANM	+			<b>←</b>	200 OK INVITE(ANM)					
			Convers	ation							
	REL	<b>→</b>			<b>→</b>	BYE(REL)					
	RLC	+			+	200 OK BYE(RLC)					

TP402005	SIP reference: RFC	3261	[4]	Q.7	-	SUP reference: Q.1912.5 [1], [i.2], clause 4.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/CLIR										
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	Restricted calling party number (user provided, not verified) Ensure that the SUT can pass transparently a call having a default calling party number with the screening indicator set to "network provided" and a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted".										
SIP parameter											
values											
ISUP parameter	IAM;										
values	Calling party number parameter  Address signals = PIXIT1  Numbering plan indicator = '001'B  Nature of address indicator = '0000011'B  Screening indicator = '11'B  Address presentation restricted parameter = '01'B  Generic number parameter  Address signals = PIXIT2  Numbering plan indicator = '001'B  Nature of address indicator = '0000011'B  Screening indicator = '00'B  Address presentation restricted parameter = '01'B										
Comments	ISUP		SUT			SIP-I					
	IAM	<b>→</b>			<del>}</del>	INVITE(IAM)					
	ACM	+			<del>-</del>	180 Ringing(ACM)					
	ANM	+			<del>-</del>	200 OK INVITE(ANM)					
			Conversa								
	REL	<b>→</b>			<del>}</del>	BYE(REL)					
	RLC	+		•	<del>(</del>	200 OK BYE(RLC)					

TSS reference   ISUP-SIP-ISUP/SS/CLIR   SIP selection criteria   ISUP selection criteria   Test purpose   Restricted calling party number (user provided, not verified) with calling sub-address   Ensure that the SUT can pass transparently a call having a default calling party numb with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted" and an access transport parameter containing the calling sub-address.  SIP parameter values   ISUP parameter   Values   IAM;   Calling party number parameter   Address signals = PIXIT1   Numbering plan indicator = '001'B   Nature of address indicator = '001'B   Address presentation restricted parameter = '01'B   Generic number parameter   Address presentation restricted parameter = '01'B   Nature of address indicator = '000'B   Nature of address indicator = '000'B   Address presentation restricted parameter = '01'B   Access transport parameter including subaddress information   Comments   ISUP	TP402006	SIP reference: RFC	3261	[4]		SUP reference: Q.1912.5 [1], [i.2], clause 4.5.2.1.1					
SIP selection criteria	TSS reference	ISUP-SIP-ISUP/SS/CLIR			Q.701	[1.2], 014436 4.5.2.1.1					
SUP selection criteria   Restricted calling party number (user provided, not verified) with calling sub-address   Ensure that the SUT can pass transparently a call having a default calling party numb with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted" and an access transport parameter containing the calling sub-address.  SIP parameter values   IAM;   Calling party number parameter   Address signals = PIXIT1   Numbering plan indicator = '001'B   Nature of address indicator = '00000011'B   Screening indicator = '11'B   Address presentation restricted parameter   Address signals = PIXIT2   Numbering plan indicator = '001'B   Nature of address indicator = '001'B   Nature of address indicator = '0000011'B   Screening indicator = '001'B   Nature of address indicator = '000'B   Address presentation restricted parameter = '01'B   Access transport parameter including subaddress information   Supplication   Supplic											
Test purpose  Restricted calling party number (user provided, not verified) with calling sub-address  Ensure that the SUT can pass transparently a call having a default calling party numb with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted" and an access transport parameter containing the calling sub-address.  SIP parameter values  IAM; Calling party number parameter Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '000'B Nature of address indicator = '000'B Nature of address indicator = '000'B Nature of address indicator = '000'B Adcress presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP  SUT  SIP-I  IAM  ACM  CM  CONVERSATION  ANM  CONVERSATION  BYE(REL)  BYE(REL)											
Test purpose  Restricted calling party number (user provided, not verified) with calling sub-address  Ensure that the SUT can pass transparently a call having a default calling party numb with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted" and an access transport parameter containing the calling sub-address.  SIP parameter values  IAM; Calling party number parameter Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '001'B Nature of address indicator = '001'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  SUP  IAM Access transport parameter including subaddress information  Comments  SUP  IAM Access transport parameter including subaddress information  Comments  SUP  IAM Access transport parameter including subaddress information  Comments  BUP  IAM Access transport parameter including subaddress information  Comments  BUP  IAM Access transport parameter including subaddress information  Comments  BUP  IAM Access transport parameter including subaddress information  Comments  BUP  IAM Access transport parameter including subaddress information  Comments  BUP  IAM Access transport parameter including subaddress information  BUP  IAM Access transport parameter including subaddress information  Comments  BUP  IAM Access transport parameter including subaddress information  BUP	ISUP selection										
sub-address  Ensure that the SUT can pass transparently a call having a default calling party numb with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted" and an access transport parameter containing the calling sub-address.  SIP parameter values  ISUP parameter values  IAM;  Calling party number parameter  Address signals = PIXIT1  Numbering plan indicator = '001'B  Nature of address indicator = '0000011'B  Screening indicator = '11'B  Address presentation restricted parameter = '01'B  Generic number parameter  Address signals = PIXIT2  Numbering plan indicator = '00001B  Nature of address indicator = '00001B  Nature of address indicator = '000011'B  Screening indicator = '00'B  Address presentation restricted parameter = '01'B  Access transport parameter including subaddress information  Comments  ISUP  SUT  SIP-I  IAM  ACM  Conversation  REL  BYE(REL)  BYE(REL)	criteria										
with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted" and an access transport parameter containing the calling sub-address.  SIP parameter values  IAM;  Calling party number parameter Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP IAM ACM FUNCTIONAM  Conversation  BYE(REL)  With the screening indicator set to "user provided, not "user provided indicator set to	Test purpose										
ISUP parameter values		with the screening indicator s additional calling party numbor verified", both having the add	e screening indicator set to "network provided", a <b>generic number</b> containing the nal calling party number with the screening indicator set to "user provided, not d", both having the address presentation restricted indicator set to "presentation"								
ISUP parameter values  IAM;  Calling party number parameter  Address signals = PIXIT1  Numbering plan indicator = '000'B  Nature of address indicator = '11'B  Address presentation restricted parameter = '01'B  Generic number parameter  Address signals = PIXIT2  Numbering plan indicator = '001'B  Nature of address indicator = '001'B  Nature of address indicator = '0000011'B  Screening indicator = '00'B  Address presentation restricted parameter = '01'B  Access transport parameter including subaddress information  Comments  ISUP  SUT  SIP-I  IAM  → INVITE(IAM)  ACM  ACM  ← 180 Ringing(ACM)  ANM  ← 200 OK INVITE(ANM)  Conversation  REL  → BYE(REL)	SIP parameter			•							
Values       Calling party number parameter         Address signals = PIXIT1       Numbering plan indicator = '0000011'B         Nature of address indicator = '11'B       Address presentation restricted parameter = '01'B         Generic number parameter       Address signals = PIXIT2         Numbering plan indicator = '001'B       Nature of address indicator = '0000011'B         Screening indicator = '00'B       Address presentation restricted parameter = '01'B         Access transport parameter including subaddress information         Comments       ISUP         IAM       → INVITE(IAM)         ACM       ← 180 Ringing(ACM)         ANM       ← 200 OK INVITE(ANM)         Conversation       → BYE(REL)	values										
Address signals = PIXIT1 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT SIP-I IAM ACM CM COnversation REL  BYE(REL)	ISUP parameter	IAM;	IAM;								
Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT SIP-I IAM ACM CM CONVERSATION ANM CONVERSATION BYE(REL)  BYE(REL)	values		neter								
Nature of address indicator = '0000011'B Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments    SUP											
Screening indicator = '11'B Address presentation restricted parameter = '01'B Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT IAM ACM ACM ANM Conversation REL  BYE(REL)  SIP-I  180 Ringing(ACM)  Conversation  BYE(REL)		,									
Address presentation restricted parameter = '01'B  Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT IAM ACM ACM ACM ANM CONVERSATION ANM ACM ACM ANM ACM ANM ACM ACM ANM ACM ACM ACM ACM ACM ACM ACM ACM ACM AC			= '0000	0011'B							
Generic number parameter Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments    SUP			_								
Address signals = PIXIT2 Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments  ISUP SUT IAM INVITE(IAM) ACM ACM ACM ANM COnversation  REL  BYE(REL)				rameter = '01'B							
Numbering plan indicator = '001'B Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments    SUP											
Nature of address indicator = '0000011'B Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments    SUP			00415								
Screening indicator = '00'B Address presentation restricted parameter = '01'B Access transport parameter including subaddress information  Comments    SUP				204415							
Address presentation restricted parameter = '01'B  Access transport parameter including subaddress information  Comments  ISUP  IAM  ACM  ACM  ANM  Conversation  REL  Address presentation restricted parameter = '01'B  SUT  SIP-I  IAM  ACM  INVITE(IAM)  ACM  Conversation  BYE(REL)			= '0000	0011'B							
Access transport parameter including subaddress information			مم امما	romotor '01'D							
ISUP   SUT   SIP-I     IAM   →   INVITE(IAM)     ACM   ←   ←   180 Ringing(ACM)     ANM   ←   ←   200 OK INVITE(ANM)     Conversation     REL   →   BYE(REL)					informati	ion					
IAM         →         INVITE(IAM)           ACM         ←         180 Ringing(ACM)           ANM         ←         200 OK INVITE(ANM)           Conversation           REL         →         BYE(REL)	Comments		JI IIICI		IIIIOIIIIat						
ACM         ←         180 Ringing(ACM)           ANM         ←         200 OK INVITE(ANM)           Conversation           REL         →         BYE(REL)	Comments		<b>→</b>	301	7						
ANM ← 200 OK INVITE(ANM)  Conversation  REL → BYE(REL)						\ /					
Conversation  REL → BYE(REL)											
REL → BYE(REL)		/ VI A161		Conversation		200 OK HAVITE(AIMI)					
		RFI	<b>→</b>	23117010411011	→	BYF(RFL)					
RIC		RLC	<del>′</del>		<del>/</del>	200 OK BYE(RLC)					

TP402007	SIP reference: RF	[4]	Q.	-	SUP reference: Q.1912.5 [1], [i.2], clause 4.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/CLIR							
SIP selection								
criteria								
ISUP selection	PICS 6/4							
criteria								
Test purpose	Discarding the calling party number if the presentation is restricted							
	Ensure that the <b>calling party number</b> is discarded in case of bilateral agreements, if the address presentation restricted indicator is set to "presentation restricted".							
SIP parameter				-				
values								
ISUP parameter	IAM;							
values	No Calling party number	parame	eter					
Comments	SIP-I		SU	Γ		ISUP		
	INVITE(IAM)	<b>→</b>		-	<b>→</b>	IAM		
	180 Ringing(ACM)	+		1	<del>(</del>	ACM		
	200 OK INVITE(ANM)	+		1	←	ANM		
			Convers	ation				
	BYE(REL) → REL							
	200 OK BYE(RLC)	+		•	<del>(</del>	RLC		

TP402008	SIP reference: RFC 3261 [4] ISUP reference:							
				•	704	Q.1912.5 [1],		
				Q	1.731	[i.2], clause 4.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/CLIR							
SIP selection								
criteria								
ISUP selection	PICS 6/4 AND PICS 6/5							
criteria								
Test purpose	Discarding the additiona	l calling	party num	ber if the	pres	sentation is restricted		
	Ensure that the additional	calling p	arty numbe	r in the ge	neric	number is discarded in case		
	of bilateral agreements, if	the addr	ess present	ation resti	ricted	indicator is set to		
	"presentation restricted".							
SIP parameter								
values								
ISUP parameter	IAM;							
values	No Calling party number	parame	eter					
	No Generic number para	meter						
Comments	SIP-I		SU	Γ		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
			Convers	ation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP402009	SIP reference: RF	C 3261	[4]		ļ	SUP reference: Q.1912.5 [1],				
				c	731	[i.2], clause 4.5.2.1.1				
TSS reference:	ISUP-SIP-ISUP/SS/CLIR					[1.2], 014400 4.0.2.111				
SIP selection criteria										
ISUP selection criteria										
Test purpose	I-MGCF: Calling party num	nber rec	eived in the	INVITE is	sent	in the IAM				
	Ensure that the calling par in the ISUP IAM.	Ensure that the calling party number contained in the encapsulated IAM is unchanged sent in the ISUP IAM.								
SIP parameter values										
ISUP parameter values										
Comments	SIP-I		SUT	•		ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	180 Ringing(ACM)	+			+	ACM				
	200 OK INVITE(ANM)	+			<b>←</b>	ANM				
			Convers	ation						
	BYE(REL)	<b>→</b>			<b>→</b>	REL				
	200 OK BYE(RLC)	+			+	RLC				

TP402010	SIP reference: RF	C 3261	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1				
TSS reference:	ISUP-SIP-ISUP/SS/CLIR						
SIP selection criteria							
ISUP selection criteria							
Test purpose	I-MGCF: Additional calling Ensure that the additional unchanged sent in the ISU	calling p					
SIP parameter values							
ISUP parameter values							
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
			Conversa	ition			
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP402011	SIP reference: RFC 3	3261	[4]	-	SUP reference:					
T00 (	Q.1912.5 [1], clause 7.1.3									
TSS reference	ISUP-SIP-ISUP/SS/CLIR									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	<ul> <li>Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is not applicable</li> <li>Sends an INVITE message with: <ul> <li>the "P-Asserted-Identity header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN;</li> <li>a "From header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN;</li> <li>and with "Privacy Header field" set to "id".</li> </ul> </li> </ul>									
SIP parameter values	INVITE: P-Asserted-Identity,	From	header field,	Privacy "id"						
ISUP parameter values	IAM: Calling party number. No	o add	itional calling	party numbe	r					
Comments	ISUP		SUT		SIP-I					
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ACM									
	ANM ← 200 OK INVITE(ANM)									
			Conversat	ion	, ,					
	REL	<b>→</b>		<b>→</b>	BYE(REL)					
	RLC	<del>(</del>		+	200 OK BYE(RLC)					

TP402012	SIP reference: RFC	3261	[4]	0.19	ISUP reference: 912.5 [1], clause 7.1.3		
TSS reference	ISUP-SIP-ISUP/SS/CLIR			Q. 1	512.5 [1], clause 1.1.5		
SIP selection criteria							
ISUP selection criteria							
Test purpose	<ul> <li>Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is applicable.</li> <li>Sends an INVITE message with:</li> <li>the "P-Asserted-Identity header field", where the user portion of the addr-spec is set to value of the calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN;</li> <li>"From header field" where the user portion of the addr-spec is set to value of the additional calling party number and the country code is set to the country where the MGCF is located in the format "+"CC+NDC+SN;</li> <li>and with "Privacy Header field" is set to "id".</li> </ul>						
SIP parameter values	INVITE: P-Asserted-Identity				'		
ISUP parameter values	IAM: Calling party number.	addition	nal calling par	ty number			
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		→	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
			Conversat	ion			
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

	Values for test purpose TP401012							
NoAS_VALUE	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 IWU 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.						

TP402013	SIP reference: RFC	3261	[4]		_	SUP reference:  2.5 [1], clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIR									
SIP selection										
criteria										
ISUP selection	PICS 1/7									
criteria										
Test purpose	Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, Privacy value "id" received.  Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation restricted.									
SIP parameter	INVITE: P-Asserted identity	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value								
values	"id" is present					•				
ISUP parameter	IAM message with the Calli									
values	Address signals = nu				sserte	ed-Identity				
	Screening indicator =									
	Number Incomplete I									
	Numbering plan indic									
	Address Presentation			itor = Pres	sentat	ion restricted				
0	NoAS: "international	numbe		-		loup				
Comments	SIP-I		SUT			ISUP				
	INVITE(IAM)	<b>→</b>			<u>→</u>	IAM				
	180 Ringing(ACM)	+			<u>+</u>	ACM				
	200 OK INVITE(ANM)	+	L		<u>←</u>	ANM				
			Convers	ation						
	BYE(REL)	<b>→</b>			<u>→</u>	REL				
	200 OK BYE(RLC)	+			+	RLC				

TP402014	SIP reference: RFC	3261	[4]		SUP reference: 12.5 [1], clause 7.1.3						
TSS reference:	ISUP-SIP-ISUP/SS/CLIR										
SIP selection											
criteria											
ISUP selection	NOT PICS 1/7										
criteria											
Test purpose	Ensure when no calling party number is included in the encapsulated IAM or the calling party number in the in the encapsulated IAM is not identical to the P-Asserted-Identity, Privacy value "id" received.  Send an IAM the calling party number is derived from SIP P-Asserted-Identity. The Address Presentation Restricted Indicator is set to Presentation restricted.										
SIP parameter	INVITE: P-Asserted identity	user p	ortion is in the	format "+"C	C+NDC+SN, Privacy value						
values	"id" is present				•						
ISUP parameter	IAM message with the Calli	ng par	ty number pa	rameter co	ded						
values	Address signals = nu Screening indicator =			IP P-Asserte	ed-Identity						
	Number Incomplete I										
	Numbering plan indic	cator =	ISDN number	ing plan							
	Address Presentation NoAS: "national (sign			= Presenta	tion restricted						
Comments	SIP-I		SUT		ISUP						
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM						
	180 Ringing(ACM)	+		<b>←</b>	ACM						
	200 OK INVITE(ANM)	+		+	ANM						
		Conversation									
	BYE(REL)	<b>→</b>		<b>→</b>	REL						
	200 OK BYE(RLC)	+		+	RLC						

TP402015	SIP reference: RFC	3261 [	4]		-	SUP reference:				
				(	Q.191	12.5 [1], clause 7.1.3				
TSS reference	ISUP-SIP-ISUP/SS/CLIR									
SIP selection										
criteria										
ISUP selection	PICS 1/7									
criteria										
Test purpose	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, Privacy value "id" received.  Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation restricted.									
SIP parameter	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value									
values	"id" is present	-				•				
ISUP parameter	IAM message with the Addi	itional	Calling par	rty numbe	er pai	rameter coded				
values	Address signals = nu					der				
	Screening indicator =			ot verified	"					
	Number Incomplete									
	Numbering plan indic									
	Address Presentation			itor = Pres	sentat	tion restricted				
	NoAS: "international	numbe		_		loup				
Comments	SIP-I	_	SU	l		ISUP				
	INVITE(IAM)	<b>→</b>			<u>→</u>	IAM				
	180 Ringing(ACM)	+			<b>←</b>	ACM				
	200 OK INVITE(ANM)	+			+	ANM				
			Convers	ation						
	BYE(REL)	<b>→</b>			<b>→</b>	REL				
	200 OK BYE(RLC)	+			+	RLC				

TP402016	SIP reference: RFC	3261	[4]			SUP reference: 2.5 [1], clause 7.1.3					
TSS reference	ISUP-SIP-ISUP/SS/CLIR										
SIP selection											
criteria											
ISUP selection	NOT PICS 1/7										
criteria											
Test purpose	Ensure when no additional calling party number is included in the encapsulated IAM or the additional calling party number in the in the encapsulated IAM is not identical to the From header field, Privacy value "id" received.  Send an IAM the additional calling party number is derived from From header field. The Address Presentation Restricted Indicator is set to Presentation restricted.										
SIP parameter	INVITE: P-Asserted identity user portion is in the format "+"CC+NDC+SN, Privacy value										
values	"id" is present										
ISUP parameter	IAM message with the Addi										
values	Address signals = nu					der					
	Screening indicator =			ot verified	"						
	Number Incomplete I										
	Numbering plan indic										
	Address Presentation			tor = Pre	sentat	ion restricted					
	NoAS: "national (sign	nificant		_		T					
Comments	SIP-I		SUT			ISUP					
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM					
	180 Ringing(ACM)	+			+	ACM					
	200 OK INVITE(ANM)	+			<b>←</b>	ANM					
			Conversa	ation							
	BYE(REL)	<b>→</b>			<b>→</b>	REL					
	200 OK BYE(RLC)	+		` <u> </u>	4	RLC					

# 5.3.3 Connected line identification presentation (COLP)

TP403001	SIP reference: RF0	C 3261	[4]	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/COLP		•					
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Initiate COLP request							
	Ensure that the exchange of the optional forward call in			fully a call	reque	esting the COLP service in		
SIP parameter values								
ISUP parameter	IAM;							
values	optional forward call indi	cators	Connected I	ine identity	/ requ	uest indicator = requested		
Comments	SIP-I		SUT	•		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	180 Ringing(ACM)	+			<del>(</del>	ACM		
	200 OK INVITE(ANM)	+			<del>(</del>	ANM		
			Conversa	ation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			<del>(</del>	RLC		

TP403002	SIP reference: RFC	3261	[4]			ISUP reference:				
					204	Q.1912.5 [1],				
T00 (					J./31	[i.2], clause 5.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/COLP									
SIP selection										
criteria ISUP selection										
criteria										
Test purpose	Connected number (user provided, verified and passed) with connected									
rest purpose	sub-address									
	Ensure that the SUT passes	transr	parently a de	efault <b>co</b> i	nnect	ed number with the				
	screening indicator set to "ve									
	containing the connected sul									
SIP parameter										
values										
ISUP parameter	IAM;									
values	optional forward call indica									
	Connected line identity reque	est ind	licator: requ	ested						
	a)									
	ANM;									
	Connected number parame		, 10	OID						
	Address presentation restrict			0.B						
	Nature of address indicator = 10 Numbering plan indicator = 10		DOTTE							
	Screening indicator = '01'B	JUID								
	Address signals = PIXIT									
	and an access transport pa	ramet	er containin	a the cor	necte	ed sub-address				
	b)		or cornaiini	9 1110 001		ou cub dudicoo.				
	CON;									
	Connected number parame	eter								
	Address presentation restrict		rameter = '0	0'B						
	Nature of address indicator =									
	Numbering plan indicator = '0	001'B								
	Screening indicator = '01'B									
	Address signals = PIXIT									
	and an access transport pa	ramet		-	necte					
Comments	SIP-I		SUT		<u> </u>	ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	CASE A		1		· •	1,014				
	180 Ringing(ACM)	<u>+</u>			<del>(</del>	ACM				
	200 OK INVITE(ANM)	+			<b>←</b>	ANM				
	CASE B									
	200 OK INVITE(CON)	+			<b>←</b>	CON				
	D)(E(DEL)		Convers	ation		251				
	BYE(REL)	<u>→</u>			<b>→</b>	REL				
	200 OK BYE(RLC)	+	ļ		+	RLC				

TP403003	SIP reference: RFC	3261	[4]	Q.73	ISUP reference: Q.1912.5 [1], 1 [i.2], clause 5.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLP		l					
SIP selection criteria								
ISUP selection								
criteria								
Test purpose	Connected number (user provided, not verified) without connected sub-address Ensure that the SUT passes transparently a default connected number with the screening indicator set to "network provided", a generic number containing the additional connected number with the screening indicator set to "user provided, not verified" without an access transport parameter containing the connected sub-address.							
SIP parameter values								
ISUP parameter	IAM;							
values	optional forward call indica	tors						
	Connected line identity reque		icator: requ	ested				
	a)	J. 1110						
	ANM;							
	Connected number parame		,	O.D.				
	Address presentation restricted Nature of address indicator =			0'B				
	Numbering plan indicator = '0		UIID					
	Screening indicator = '11'B							
	Address signals = PIXIT							
	Additional connected numb			OID				
	Address presentation restricted Nature of address indicator =			0.B				
	Numbering plan indicator = '0		0116					
	Screening indicator = '00'B	,0.5						
	Address signals = PIXIT							
	b)							
	CON;							
	Connected number parame			0.I <b>D</b>				
	Address presentation restricted Nature of address indicator =			0.B				
	Numbering plan indicator = '0		0116					
	Screening indicator = '11'B	,0.5						
	Address signals = PIXIT							
	Additional connected number			0.I <b>D</b>				
	Address presentation restricted Nature of address indicator =			0.B				
	Numbering plan indicator = '0		UIID					
	Screening indicator = '00'B							
	Address signals = PIXIT							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	CASE A	<del>-</del>		<b>+</b>	IACM			
	180 Ringing(ACM) 200 OK INVITE(ANM)	<del>-</del>		+	ACM ANM			
	CASE B				į, arvivi			
	200 OK INVITE(CON)	<del>(</del>		+	CON			
			Convers	ation				
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	<del>-</del>		+	RLC			

TP403004	SIP reference: RFC	3261	[4]	Q.7:	Q.19	reference: 912.5 [1], clause 5.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/COLP										
SIP selection											
criteria											
ISUP selection	PICS 1/7										
criteria											
Test purpose	Converting the connected number to national format, if necessary  Ensure that the country code in the address signals of the connected number is removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number", the address presentation restricted indicator and the screening indicator shall be transferred transparently.										
SIP parameter	200 OK: encapsulated ANM										
values	Connected number										
	Address presentation										
	Nature of address inc Numbering plan indic			В							
	Screening indicator =										
	Address signals = PI		_0.								
ISUP parameter	IAM;										
values	optional forward call indic	ators									
	Connected line identity requ		icator: requi	ested							
	a)	iest iiiu	icator. requ	esteu							
	ANM;										
	Connected number param	eter									
	Address presentation restric	cted par	rameter = '0	0'B							
	Nature of address indicator		)100'B								
	Numbering plan indicator =										
	Screening indicator = ISUP_										
	Address signals = CC+PIXI	ı									
	b) CON;										
	Connected number param	eter									
	Address presentation restrict		rameter = '0	0'B							
	Nature of address indicator										
	Numbering plan indicator =	'001'B									
	Screening indicator = ISUP_										
	Address signals = CC+PIXI										
Comments	Generic number paramete	r not p		-	licur						
Comments		<b>→</b>	SUT	-	ISUF IAM						
	INVITE(IAM)  CASE A	7		7	IAIVI						
	180 Ringing(ACM)	+		•	- ACM	1					
	200 OK INVITE(ANM)	+									
	CASE B	_	I		7 (1410	•					
	200 OK INVITE(CON)	+		•	- CON	J					
			Convers		331	-					
	BYE(REL)	<b>→</b>		-	REL						
	200 OK BYE(RLC)	<b>←</b>		•							

TP403005	SIP reference: RFC 32	261 [4	ı]		ISUP reference:			
				0.73	Q.1912.5 [1], 31 [i.2], clause 5.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLP			4.1.	71 [112], 010000 010121111			
SIP selection	1001 011 1001 70070021							
criteria								
ISUP selection	PICS 1/7							
criteria								
Test purpose	Converting the additional connected number to national format, if necessary							
	Ensure that the country code in the address signals of the <b>generic number</b> coded as an							
	"additional connected number", if the numbering plan indicator is "ISDN Telephony" is							
	removed if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number", the address presentation restricted indicator and the							
	screening indicator shall be transferred transparently.							
SIP parameter	200 OK: encapsulated ANM or CON							
values	additional connected number							
	Address presentation restricted parameter = '00'B							
	Nature of address indicator = '0000011'B							
	Numbering plan indicator = '001'B							
	Screening indicator = '01'B							
	Address signals = PIXIT							
ISUP parameter	IAM;							
values	optional forward call indicators							
	Connected line identity request indicator: requested							
	a)							
	ANM;							
	Connected number parameter present							
	additional connected number							
	Address presentation restricted parameter = '00'B  Nature of address indicator = '000100'B							
	Numbering plan indicator = '000'B							
	Screening indicator = '01'B							
	Address signals = CC+PIXIT							
	b)							
	CON;							
	Connected number parameter present							
	additional connected number							
	Address presentation restricted parameter = '00'B							
	Nature of address indicator = '000100'B							
	Numbering plan indicator = '001'B							
	Screening indicator = '01'B Address signals = CC+PIXIT							
Comments	SIP-I		SUT		ISUP			
		<b>&gt;</b>		<b>→</b>				
	CASE A	<u> </u>		<u> </u>	•			
		<b>-</b>		+	ACM			
	200 OK INVITE(ANM)	<b>-</b>		+				
	CASE B			-				
	200 OK INVITE(CON)	<b>E</b>		+	CON			
			Conversa					
		<b>→</b>		<b>→</b>				
	200 OK BYE(RLC)	<b>-</b>		+	RLC			

TP403006	SIP reference: RFC	3261 [4]	Q.73	ISUP reference: Q.1912.5 [1], 1 [i.2], clause 5.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/COLP							
SIP selection								
criteria ISUP selection	DIOC 4/0 AND DIOC 7/5							
criteria	PICS 1/8 AND PICS 7/5							
Test purpose	Adding a prefix to an international connected number							
rest purpose	Ensure that a prefix is added to the <b>connected number</b> and the nature of address							
	indicator is set to "unknown" (see note).							
SIP parameter	200 OK INVITE with encapsulated ANM or CON							
values	Connected number parameter							
	Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT							
ISUP parameter	ANM/CON:							
values	Connected number parameter							
	Address presentation restricted parameter = '00'B							
	Nature of address indicator = '0000010'B							
	Numbering plan indicator = '001'B							
	Screening indicator = '11'B							
	Address signals = Prefix+PIXIT							
Comments	SIP-I		SUT	ISUP				
	IAM	<b>→</b>	→	INVITE(IAM)				
	CASE A							
	ACM	<del>-</del>	<b>←</b>	180 Ringing(ACM)				
	ANM	<del>-</del>	<b>←</b>	200 OK INVITE(ANM)				
	CASE B							
	CON	+	+	200 OK INVITE(CON)				
		Conve	ersation					
	REL	<b>→</b>	→	BYE(REL)				
	RLC	+	+	200 OK BYE(RLC)				
NOTE: The cod	ing "unknown" is a national op	tion (@).		. ,				

TP403007	SIP reference: RFC 32	261 [4]		SUP reference: Q.1912.5 [1], [i.2], clause 5.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLP						
SIP selection criteria							
ISUP selection criteria	PICS 1/8 AND PICS 7/3						
Test purpose	Discarding the connected nu Ensure that the connected nu address presentation restricted	mber is discard	ded in case of b	ilateral agreements, if the			
SIP parameter	200 OK INVITE with encapsula			(-1			
values	Connected number pa						
	Address presentation restricted parameter = '00'B Nature of address indicator = '0000011'B Numbering plan indicator = '001'B Screening indicator = '11'B Address signals = PIXIT						
ISUP parameter	IAM						
values	optional forward call indicate Connected line identity reques a) ANM No Connected number parar b) CON; No Connected number parar	t indicator: requ neter neter	_				
Comments	ISUP	SU'	Т	SIP-I			
	IAM -	<b>&gt;</b>	<b>→</b>	INVITE(IAM)			
	CASE A						
		<del>-</del>	<del>-</del>	180 Ringing(ACM)			
	ANM	-	<b>+</b>	200 OK INVITE(ANM)			
	CASE B						
	CON	-	<b>+</b>	200 OK INVITE(CON)			
		Convers					
	REL -		<b>→</b>	BYE(REL)			
	ILLO	=	<b>+</b>	200 OK BYE(RLC)			
	teral agreement prohibits the trar ess presentation restricted indica						

TP403008	SIP reference: RFC 32	61 [4]		ISUP reference: Q.1912.5 [1],				
			Q 731	[i.2], clause 5.5.2.1.1				
TSS reference:	ISUP-SIP-ISUP/SS/COLP		Q.,, O.1	[2], 0.0000 0.0.2.111				
SIP selection	1001 011 1001 700/0021							
criteria								
ISUP selection	PICS 1/8 AND PICS 7/4							
criteria								
Test purpose	Discarding the additional connected number in case of bilateral agreements							
		Ensure that the additional connected number in the <b>generic number</b> is discarded in case						
	of bilateral agreements, if the a		tation restricted	l indicator is set to				
	"presentation allowed" (see not							
SIP parameter	200 OK INVITE with encapsula							
values	Additional Connected							
	Address presentation re							
	Nature of address indicate		ΙΒ					
	Numbering plan indicate Screening indicator = '0							
	Address signals = PIXIT							
ISUP parameter	The second secon							
values	IAM;							
14.000	optional forward call indicators							
	Connected line identity request	indicator: requ	ıested					
	a)							
	ANM;							
	No Connected number parame	ter						
	No Additional connected numb	er present						
	b)							
	CON;							
	No Connected number parame							
Cammanta	No Additional connected numb	er present SU	<del>-</del>	SIP-I				
Comments	IAM =		· →	INVITE(IAM)				
	CASE A		7	INVITE(IAIVI)				
	ACM (	<u>.</u>	+	180 Ringing(ACM)				
	ANM		+	200 OK INVITE(ANM)				
	CASE B	-		200 OK INVITE(ANVI)				
	CON	<b>-</b>	+	200 OK INVITE(CON)				
		Convers		200 010 110011 (0010)				
	REL -		<b>→</b>	BYE(REL)				
	RLC •		<del>,</del>	200 OK BYE(RLC)				
NOTE: This bila	teral agreement prohibits the tran							
	in any case.		aditional confine	otos nambor in the generic				
Hamber	, 0							

TP403009	SIP reference: RFC	3261	[4]		ı	SUP reference:
						Q.1912.5 [1],
				G	2.731	[i.2], clause 5.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLP					
SIP selection						
criteria						
ISUP selection	PICS 1/8					
criteria						
Test purpose	Converting the connected					
	Ensure that the exchange ca					
						nal number" and can pass on
CID noromotor	the address presentation re-				reenir	ng indicator transparently.
SIP parameter values	200 OK INVITE with encaps			N		
values	Connected number Address presentation			tor - '00'	D	
	Nature of address in				Ь	
	Numbering plan indic			Ь		
	Screening indicator =		001 B			
	Address signals = Co		Т			
ISUP parameter	IAM;					
values	optional forward call indic	ators				
	Connected line identity requ	est ind	icator: reque	ested		
	a)		•			
	ANM					
	Connected number param					
	Address presentation restrict			0'B		
	Nature of address indicator		)100'B			
	Numbering plan indicator =	'001'B				
	Screening indicator = '11'B					
	Address signals = PIXIT Presentation restricted indic	otor – '	יחחים			
	additional connected num					
	b)	Dei pi	536111			
	CON;					
	Connected number param	eter				
	Address presentation restrict		rameter = '0	0'B		
	Nature of address indicator					
	Numbering plan indicator =	'001'B				
	Screening indicator = '11'B					
	Address signals = PIXIT					
	Presentation restricted indic					
	additional connected num	<b>ber</b> pre		,		lious
Comments	SIP-I		SUT		-	ISUP
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM
	CASE A		ı			IA ONA
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
	CASE B	L			<i>I</i> _	CON
	200 OK INVITE(CON)	+	Carre		+	CON
	DVE(DEL)	_	Conversa	ation		DEL
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	+			+	RLC

TP403010	SIP reference: RF	C 3261	[4]		ISUP reference: Q.1912.5 [1], [i.2], clause 5.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLP							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Handling unrequested Confidence that the call can be		sfully set up	if the SUT red	ceives an unsolicited COL.			
SIP parameter	200 OK INVITE with encar	sulated	ANM or CO	N				
values	Connected number							
	Address presentation							
	Nature of address in	ndicator	= '0000011'	В				
	Numbering plan ind		'001'B					
	Screening indicator							
	Address signals = F	PIXIT						
ISUP parameter	IAM;							
values	optional forward call indi							
	Connected line identity request indicator: not requested							
	(a)							
	ANM;							
	Connected number parameter							
	Address presentation restr			0'B				
	Nature of address indicato		0011'B					
	Numbering plan indicator =							
	Screening indicator = '11'B	3						
	Address signals = PIXIT							
	additional connected nui	mber pr	esent					
	b) <b>CON</b> ;							
	Connected number parameter							
	Address presentation restricted parameter = '00'B							
	Nature of address indicator = '0000011'B							
	Numbering plan indicator = '001'B							
	Screening indicator = '11'B Address signals = PIXIT							
	additional connected nu	mher nr	acant					
Comments	SIP-I	Tibel pi	SUT		ISUP			
Comments	INVITE(IAM)	<b>→</b>	301	<b>→</b>	IAM			
	CASE A	7	1	-	IIVIAI			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		<del></del>	ANM			
	CASE B			<u> </u>	AINIVI			
					CON			
	200 OK INVITE(CON)	+	Carriaria	← tion	CON			
	DVE(DEL)	+	Conversa		DEL			
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		←	RLC			

TP403012	SIP reference: RFC	3261 [4	4]		ISUP reference:		
					Q.1912.5 [1],		
				Q.731	[i.2], clause 5.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/CLIR						
SIP selection							
criteria	2122 1/2						
ISUP selection	PICS 1/7						
criteria	F			000 01/ 1811//	TE is south and the TOLID side		
Test purpose	without changing. The conn				TE is sent on the ISUP side		
	connected sub address is in			changed. The	ATP contained the		
SIP parameter	200 OK INVITE: encapsulat			sluded			
values	200 OK INVITE. encapsulat	leu Aiviv	I OI CON III	Juueu			
ISUP parameter	a)						
values	ANM;						
	Connected number param	eter					
	Address presentation restrict		ameter = '00	)'B			
	Nature of address indicator						
	Numbering plan indicator =	'001'B					
	Screening indicator = '11'B						
	Address signals = PIXIT						
	and an access transport pa	aramete	er containing	the connecte	ed sub-address.		
	b)						
	CON;						
	Connected number param						
	Address presentation restrict			)'B			
	Nature of address indicator		011'B				
	Numbering plan indicator =	001B					
	Screening indicator = '11'B Address signals = PIXIT						
	and an access transport pa	aramete	ar containing	the connect	ad sub-address		
Comments	ISUP		SUT	THE COMPLECT	SIP-I		
Commonto	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	CASE A						
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
	CASE B				1=== = = = = = = = = = = = = = = = = =		
	CASE B  CON ←   200 OK INVITE(CON)						
1	CON				IZUU UN INVITETUUNI		
1	CON		Conversa	_	200 OK INVITE(CON)		
	REL	<b>→</b>	Conversa	_	BYE(REL)		

TP403013	SIP reference: RFC 3261 [4]			SUP reference: Q.1912.5 [1], [i.2], clause 5.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/CLIR	L		<u></u>			
SIP selection	100. 0 100. 700, 02						
criteria							
ISUP selection							
criteria							
Test purpose	O-MGCF: connected number and add	litional connecte	ed numb	ber transferred transparently			
	Ensure that an ANM or CON encapsu without changing. The connected num connected sub address is included.						
SIP parameter values	200 OK INVITE: encapsulated ANM o	r CON included	1				
ISUP parameter	a)						
values	ANM;						
	Connected number parameter						
	Address presentation restricted param	neter = '00'B					
	Nature of address indicator = '000001	1'B					
	Numbering plan indicator = '001'B						
	Screening indicator = '11'B						
	Address signals = PIXIT						
	Additional connected number prese						
	Address presentation restricted param						
	Nature of address indicator = '000001	1'B					
	Numbering plan indicator = '001'B						
	Screening indicator = '00'B						
	Address signals = PIXIT						
	and an access transport parameter of	containing the c	onnecte	ed sub-address.			
	b)						
	CON;						
	Connected number parameter						
	Address presentation restricted param						
	Nature of address indicator = '000001	1'B					
	Numbering plan indicator = '001'B						
	Screening indicator = '11'B						
	Address signals = PIXIT	.nt					
	Additional connected number prese						
	Address presentation restricted param Nature of address indicator = '000001						
	Numbering plan indicator = '001'B	טו					
	Screening indicator = '00'B						
	Address signals = PIXIT						
	and an access transport parameter of	containing the c	onnecte	ed sub-address			
Comments	ISUP	SUT		SIP-I			
- January 1113	IAM →	<del> </del>	→	INVITE(IAM)			
	CASE A						
	ACM ←		+	180 Ringing(ACM)			
	ANM ←		+	200 OK INVITE(ANM)			
	CASE B			1200 OK HAVITE (AINIVI)			
	CON +		+	200 OK INVITE(CON)			
		Conversation		ZOU OR INVITE(CON)			
		onversation		DVE(DEL)			
	REL →		<b>→</b>	BYE(REL)			
	RLC ←			200 OK BYE(RLC)			

### 5.3.4 Connected Line Identification Restriction (COLR)

TP404001	SIP reference: RFC	3261	[4]	I	SUP reference: Q.1912.5 [1],
				Q.731	[i.2], clause 6.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLR			4	[],
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Passing on information re				
	Ensure that the SUT shall p				
	supplementary service in the	e addre	ess presentation res	tricted	indicator of the connected
0.0	number.				
SIP parameter					
values	LOBA.				
ISUP parameter values	IAM; optional forward call indic	notoro			
values	Connected line identity requ		liantar: raquantad		
	a)	Jest ind	licator, requested		
	ANM;				
	Connected number paran	neter			
	Address presentation restrict		rameter = '01' B		
	Nature of address indicator				
	Numbering plan indicator =	'001'B			
	Screening indicator = '01'B				
	Address signals = PIXIT				
	b)				
	CON;				
	Connected number paran				
	Address presentation restrict				
	Nature of address indicator		0011'B		
	Numbering plan indicator = Screening indicator = '01'B	001B			
	Address signals = PIXIT				
	Address signals = FIXIT				
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	CASE A	1	1	-	ı
	180 Ringing(ACM)	<b>←</b>		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
	CASE B	•		•	
	200 OK INVITE(CON)	<b>←</b>		+	CON
			Conversation		
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP404002	SIP reference: RFC 3	261 [4	]		ISUP reference: Q.1912.5 [1], [i.2], clause 6.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLR				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Passing on information related Ensure that the SUT shall pass supplementary service in the anumber and the additional control of the supplementary service.	s trans addres	sparently all s presentat	ion restricted	indicator of the connected
SIP parameter					
values					
ISUP parameter values	IAM; optional forward call indicat Connected line identity reques a)		ator: reque	sted	
	ANM;				
	Connected number paramet		motor – '04	' D	
	Address presentation restricte Nature of address indicator = 1			В	
	Numbering plan indicator = '00		IIID		
	Screening indicator = '11'B	010			
	Address signals = PIXIT				
	Additional connected number	er pres	sent		
	Address presentation restricte	-		' B	
	Nature of address indicator = '				
	Numbering plan indicator = '00	01'B			
	Screening indicator = '00'B				
	Address signals = PIXIT				
	b)				
	CON;				
	Connected number paramet			ı D	
	Address presentation restricte	•		В	
	Nature of address indicator = '		ПБ		
	Numbering plan indicator = '00 Screening indicator = '11'B	υιь			
	Address signals = PIXIT				
	Additional connected number	er pres	sent		
	Address presentation restricte			' B	
	Nature of address indicator = '				
	Numbering plan indicator = '00				
	Screening indicator = '00'B				
	Address signals = PIXIT				<del>_</del>
Comments	SIP-I		SUT		ISUP
	\ /	→		→	IAM
	CASE A			1	T
	5 5 7	<del>(</del>		<del>-</del>	ACM
	,	<b>←</b>		<del>(</del>	ANM
	CASE B	<del>-</del> 1			laav.
	200 OK INVITE(CON)	<b>←</b>		<u> </u>	CON
			Conversat	_	
		<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	<del>(</del>		+	RLC

TP404003	SIP reference: RFC	3261	[4]		ISUP reference:			
				0.73	Q.1912.5 [1], 1 [i.2], clause 6.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLR			Q.73	1 [1.2], Clause 0.3.2.1.1			
SIP selection	1301 -311 -1301 /33/COLIX							
criteria								
ISUP selection								
criteria								
Test purpose	Restricted connected number (user provided, verified and passed) with connected sub-address							
	Ensure that the SUT can pas							
	indicator set to "user provide							
	restricted indicator set to "pr							
	Additionally, an access tran shall also be provided.	sport	parameter	containing the	connected sub-address			
SIP parameter	janan also be provided.							
values								
ISUP parameter	IAM;							
values	optional forward call indic	ators						
	Connected line identity requ		icator: requ	ested				
	a)		·					
	ANM;							
	Connected number parame							
	Address presentation restric			)1' B				
	Nature of address indicator		0011'B					
	Numbering plan indicator = '	001'B						
	Screening indicator = '01'B Address signals = PIXIT							
	access transport parameter	contair	ning the cor	nacted sub-a	ddrees			
	b)	Contail	iiig the cor	inected Sub-a	duless			
	CON:							
	Connected number parame	eter						
	Address presentation restric	ted pai	rameter = '0	)1' B				
	Nature of address indicator :	= '0000	011'B					
	Numbering plan indicator = '	001'B						
	Screening indicator = '01'B							
	Address signals = PIXIT				44			
Comments	access transport parameter SIP-I	contair	su		ISUP			
Comments	INVITE(IAM)	<b>→</b>	30	· →	IAM			
	CASE A	7		7	IIVIAI			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	<del>`</del>		+	ANM			
	CASE B				l			
	200 OK INVITE(CON)	+		+	CON			
			Convers	ation				
	BYE(REL)	<b>→</b>		→	REL			
	200 OK BYE(RLC)	+		+	RLC			
	( / )				<u> </u>			

TP404004	SIP reference: RFC	3261	[4]		I	SUP reference: Q.1912.5 [1],
				Q.7	31	[i.2], clause 6.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/COLR					<u></u>
SIP selection						
criteria						
ISUP selection	PICS 7/1					
criteria						
Test purpose	Discarding the connected	numb	er if the pre	sentation i	is re	estricted
	Ensure that the connected					
	address presentation restric			to "presenta	atio	n restricted".
SIP parameter	200 INVITE: encapsulated A					
values	No Connected numb	er para	ameter inclu	ded		
ISUP parameter	IAM;					
values	optional forward call indic					
	Connected line identity requ	est ind	icator: requ	ested		
	a)					
	ANM;					
	Connected number param	eter		AID		
	Address presentation restric			T.B		
	Nature of address indicator		DOTTE			
	Numbering plan indicator = ' Screening indicator = '11'B	ООГБ				
	Address signals = PIXIT					
	Address signals = 1 IXII					
	b)					
	CON;					
	Connected number param	eter				
	Address presentation restrict		rameter = '0	1'B		
	Nature of address indicator			-		
	Numbering plan indicator = '					
	Screening indicator = '11'B					
	Address signals = PIXIT					
Comments	SIP-I		SUT	•		ISUP
	INVITE(IAM)	<b>→</b>		-	•	IAM
	CASE A					
	180 Ringing(ACM)	+		•		ACM
	200 OK INVITE(ANM)	+		•		ANM
	CASE B					
	200 OK INVITE(CON)	+		•		CON
			Convers			
	BYE(REL)	<b>→</b>		-		REL
	200 OK BYE(RLC)	+		•	_	RLC

TP404005	SIP reference: RFC 3	3261 [4]			SUP reference: Q.1912.5 [1], [i.2], clause 6.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLR				,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,,			
SIP selection	PICS 7/2							
criteria								
ISUP selection								
criteria								
Test purpose	Discarding the additional co	onnecte	d numbe	r in the gene	ric number if the			
	presentation is restricted							
	Ensure that the additional cor							
	of bilateral agreements, if the presentation restricted.	address	presenta	mon restricted	indicator is set to			
SIP parameter	200 INVITE: encapsulated AN	VM or CC	)N					
values	No Additional Connect			neter included				
ISUP parameter	IAM;	tou manni	bor paran	iotor irroradoa				
values	optional forward call indica	tors						
	Connected line identity reque		tor: reque	ested				
	a)		•					
	ANM;							
	Connected number paramet							
	Additional Connected number			415				
	Address presentation restricted Nature of address indicator =			I.R				
	Numbering plan indicator = '0		ID					
	Screening indicator = '11'B	010						
	Address signals = PIXIT							
	<b>L</b>							
	b) CON;							
	Connected number paramet	ter nrese	nt					
	Additional Connected number							
	Address presentation restricted			1'B				
	Nature of address indicator = '0000011'B							
	Numbering plan indicator = '0	01'B						
	Screening indicator = '11'B							
	Address signals = PIXIT							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		→	IAM			
	CASE A							
	180 Ringing(ACM)	<del>(</del>		<del>-</del>	ACM			
	200 OK INVITE(ANM)	<b>←</b>		<b>+</b>	ANM			
	CASE B			- بر	loon			
	200 OK INVITE(CON)	<del>-</del>	<b>^</b>	<b>(</b>	CON			
	DVE(DEL)		Conversa		DEL			
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	<del>-</del>		<b>←</b>	RLC			

TP404007	SIP reference: RFC 3261 [4	1]	ı	SUP reference:			
			Q 731	Q.1912.5 [1], [i.2], clause 6.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/COLR	J	Q./ 0 1	[112], 014400 010.21111			
SIP selection	100. 0 100. 700, 00 2						
criteria							
ISUP selection							
criteria							
Test purpose	O-MGCF: Connected number, addit	tional connec	ted number	and connected subaddress			
	transferred						
	Ensure that an ANM or CON encaps						
	without changing. The connected nu	ımber is unch	nanged. The	ATP contained the			
OID	connected sub address is included.						
SIP parameter values	200 OK INVITE: encapsulated ANM	or CON incl	uded				
ISUP parameter	ANM;						
values	Connected number parameter						
	Address presentation restricted para	ameter = '01'	В				
	Nature of address indicator = '00000						
	Numbering plan indicator = '001'B						
	Screening indicator = '11'B						
	Address signals = PIXIT						
	Additional connected number pre-						
	Address presentation restricted para		В				
	Nature of address indicator = '00000	011'B					
	Numbering plan indicator = '001'B						
	Screening indicator = '00'B						
	Address signals = PIXIT	r containing	ha aannaata	ad out oddroog			
	and an access transport paramete	r containing	ine connecte	ed Sub-address.			
	b) CON;						
	Connected number parameter						
	Address presentation restricted para	ameter = '01'	В				
	Nature of address indicator = '00000						
	Numbering plan indicator = '001'B						
	Screening indicator = '11'B						
	Address signals = PIXIT						
	Additional connected number present						
	Address presentation restricted parameter = '01'B						
	Nature of address indicator = '00000	)11'B					
	Numbering plan indicator = '001'B						
		Screening indicator = '00'B					
	Address signals = PIXIT			ad a culto a adalas a a			
Cammanta	and an access transport paramete		ine connecte				
Comments	ISUP IAM →	SUT	<b>→</b>	SIP-I INVITE(IAM)			
	IAM →		7	IIIVII E(IAWI)			
	ACM ←		+	180 Ringing(ACM)			
	ANM		<del>-</del>	200 OK INVITE(ANM)			
	CASE B		•	ZOO ON INVITE(ANIVI)			
	CON +		+	200 OK INVITE(CON)			
	CON	Conversati		ZUU UK IINVI I E(UUIN)			
	REL →	Conversati	on →	BYE(REL)			
	RLC +		+	200 OK BYE(RLC)			
	INLO			1200 ON DIL(NLO)			

### 5.3.5 Terminal Portability (TP)

TP405001	SIP reference:	RFC 3261	[4]		SUP reference: Q.1912.5 [1], 3 [i.6], clause 4.5.2.1
TSS reference:	ISUP-SIP-ISUP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Terminal portability, re Ensure that SUT inform requested by the calling	s the called	party that a si	uspend and	a resume have been US and RES messages.
SIP parameter					osulated in the MIME body
values					-
ISUP parameter					
values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	1		+	200 OK INVITE(ANM)
			Conversation	on	
	SUS	<b>→</b>		<b>→</b>	INFO(SUS)
				+	200 OK INFO
	RES	<b>→</b>		<b>→</b>	INFO(RES)
				+	200 OK INFO
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP405002	SIP reference	e: RFC 3261	[4]		ISUP reference: Q.1912.5 [1], 3 [i.6], clause 4.5.2.1
TSS reference:	ISUP-SIP-ISUP/SS/T	TP			
SIP selection criteria					
ISUP selection criteria					
Test purpose	requested by the call	rms the calling ed party upon	party that a su receipt of user	ispend and initiated <b>S</b>	d a resume have been US and RES messages.
SIP parameter	INFO: Content-Type:	application/IS	UP ; SUS and	RES enca	psulated in the MIME body
values					
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversatio	n	
	SUS	+		+	INFO(SUS)
				<b>→</b>	200 OK INFO
	RES	+		+	INFO(RES)
			-	<b>→</b>	200 OK INFO
	DEL				D)(E(DEL)
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	<b>←</b>		+	200 OK BYE(RLC)

TP405003	SIP reference:	RFC 3261 [	4]	Q.73	ISUP reference: Q.1912.5 [1], 33 [i.6], clause 4.5.2.1			
TSS reference	ISUP-SIP-ISUP/SS/TP							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that the call is re	Terminal portability, requested by local served user, no Resume after Suspend Ensure that the call is released with cause #102 (recovery on timer expiry) by the SUT if timer T2 expires because the local served user does not resume the call.						
SIP parameter values	INFO: Content-Type: ap BYE: Content-Type: ap	plication/IS	UP ; SUS e	ncapsulated	in the MIME body			
ISUP parameter values		•	·	•	•			
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversa	tion				
	SUS	SUS → INFO(SUS)						
				+	200 OK INFO			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		<del>-</del>	200 OK BYE(RLC)			

TP405004	SIP reference: RFC	3261	[4]	_	SUP reference: Q.1912.5 [1], 3 [i.6], clause 4.5.2.1			
TSS reference	ISUP-SIP-ISUP/SS/TP							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Terminal portability, relea	se sus	pended call					
	Ensure that a suspended ca	all can b	e released, if	the remote ι	user releases the call.			
SIP parameter	INFO: Content-Type: applic	ation/IS	SUP ; SUS end	apsulated ir	the MIME body			
values	BYE : Content-Type: application	ation/IS	UP; REL enca	apsulated in	the MIME body			
ISUP parameter								
values								
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
	Conversation							
	SUS → INFO(SUS)							
					200 OK INFO			
	REL	+		+	BYE(REL)			
	RLC	<b>→</b>		<b>→</b>	200 OK BYE(RLC)			

# 5.3.6 SUB-addressing (SUB)

TP406001	SIP reference: RFC	3261	[4]	_	SUP reference: Q.1912.5 [1], [i.2], clause 8.5.2.1.1/			
TSS reference:	ISUP-SIP-ISUP/SS/SUB							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that the SUT can incomparameter in the encapsula	clude th	re called sub-add	,				
SIP parameter values	INVITE: Content-Type: app	lication	/ISUP ; IAM enca	psulated	in the MIME body			
ISUP parameter values								
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM ← 200 OK INVITE(ANM)							
			Conversation					
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP406002	SIP reference: RF	C 3261	[4]	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 8.5.2.1.1/					
TSS reference	ISUP-SIP-ISUP/SS/SUB								
SIP selection criteria									
ISUP selection criteria									
Test purpose		Receiving the called sub-address in the access transport parameter							
	Ensure that the SUT can in parameter in the ISUP IAM		ne called sur	-address in t	ne access transport				
SIP parameter values									
ISUP parameter									
values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversa	ıtion					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP406003	SIP reference: RFC	3261	[4]	-	SUP reference: Q.1912.5 [1], [i.2], clause 8.5.2.1.1/			
TSS reference	ISUP-SIP-ISUP/SS/SUB							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that the SUT can incorparameter in the encapsular	clude th	ne calling sub-ado					
SIP parameter values	INVITE: Content-Type: appl			psulated	in the MIME body			
ISUP parameter values								
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM ← 200 OK INVITE(ANM)							
			Conversation	•				
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP406004	SIP reference: RI	FC 3261	[4]		SUP reference: Q.1912.5 [1], [i.2], clause 8.5.2.1.1/				
TSS reference	ISUP-SIP-ISUP/SS/SUB								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT can	Receiving the calling sub-address in the access transport parameter  Ensure that the SUT can include the calling sub-address in the access transport parameter in the ISUP IAM.							
SIP parameter values									
ISUP parameter values									
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		→	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversation	on					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

### 5.3.7 Malicious Call Identification (MCID)

TP407001	SIP reference: RFC	3261	[4]	Q.73	ISUP reference: Q.1912.5 [1], 1.7 [i.3], clause 7.5.2.1.1		
TSS reference:	ISUP-SIP-ISUP/SS/MCID						
SIP selection criteria							
ISUP selection criteria							
Test purpose  SIP parameter	Successful MCID request O-MGCF  Ensure that the SUT can successfully pass on a 183 Session Progress containing an encapsulated IDR having the MCID request indicator set to "MCID request" and pass on an IRS with MCID response indicator set to "MCID included" and the calling party number included. ISUP to SIP-I interworking.  183 Session Progress: Content-Type: application/ISUP; IDR encapsulated in the MIME						
values	body	iterit- i y	pe. application	1/15UP, 1D	R encapsulated in the MilME		
Value	INFO: Content-Type: applic	ation/IS	SUP: IRS enca	apsulated i	n the MIME body		
ISUP parameter values			,	•	,		
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	IDR	+		+	183 Session Progress(IDR)		
	IRS	<b>→</b>		<b>→</b>	INFO(IRS)		
				+	200 OK INFO		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
			Conversation	on			
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP407002	SIP reference: RFC 32	261 [4]			SUP reference: Q.1912.5 [1], 7 [i.3], clause 7.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/MCID								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT can successet to "MCID request" and pas	Successful MCID request I-MGCF  Ensure that the SUT can successfully pass on an IDR having the MCID request indicator set to "MCID request" and pass on an IRS with MCID response indicator set to "MCID included" and the calling party number included. SIP-I to ISUP interworking.							
SIP parameter values	183 Session Progress: Content body INFO: Content-Type: application				•				
ISUP parameter values									
Comments	SIP-I		SU <sup>*</sup>	Γ	ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	183 Session Progress(IDR)	+		+	IDR				
	INFO(IRS)	<b>→</b>		<b>→</b>	IRS				
	200 OK INFO	+							
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Convers	ation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	<b>+</b>		+	RLC				

TP407003	SIP reference: RFC 3261 [4]			Q.		ISUP reference: Q.1912.5 [1], 7 [i.3], clause 7.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/MCID					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Successful MCID request - at					
	Ensure that the SUT will accep					
	been received. The SUT should					
	"MCID request" and pass on ar					
	included" and the calling party					
SIP parameter	INFO: Content-Type: application					
values	INFO: Content-Type: application				d in	the MIME body
ISUP parameter	IRS containing the calling party	numbe	r paran	neter		
values		1				T
Comments	SIP-I		S	UT		ISUP
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM
	CASE A	•				
	180 Ringing(ACM)	+			<b>←</b>	ACM
	183 Session Progress(IDR)	+			+	IDR
	INFO(IRS)	<b>→</b>			<b>→</b>	IRS
	200 OK INFO	<b>←</b>				
	200 OK INVITE(ANM)	+			+	ANM
	CASE B					
	183 Session Progress(ACM)	+			+	ACM(early)
	183 Session Progress(IDR)	+			+	IDR
	INFO(IRS)	<b>→</b>			<b>→</b>	IRS
	200 OK INFO	+				
	180 Ringing(CPG)	+			+	CPG(alerting)
	200 OK INVITE(ANM)	+			+	ANM
	` '		Conve	rsation		
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	+			+	RLC
NOTE: This situa	ation may occur e.g. if the call ha	s been	forward	ed before	reac	hing the destination.

TP407004	SIP reference: RF	C 3261	[4]	Q.7:	ISUP reference: Q.1912.5 [1], 31.7 [i.3], clause 7.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/MCID							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Ensure that the SUT rejec	MCID request - MCID not supported by the OLE O-MGCF Ensure that the SUT rejects a MCID request by sending an IRS with the MCID response indicator set to "MCID not included". ISUP to SIP-I interworking.						
SIP parameter					OR encapsulated in the MIME			
values	body INFO: Content-Type: appl	-			•			
ISUP parameter values				•	·			
Comments	ISUP		SUT		SIP-I			
	IAM	→		<b>→</b>	INVITE(IAM)			
	IDR	+		+	183 Session Progress(IDR)			
	IRS	<b>→</b>		<b>→</b>	INFO(IRS)			
				+	200 OK INFO			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversa	tion				
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP407005	SIP reference: RFC 32	261 [4]		ISUP reference: Q.1912.5 [1], Q.731.7 [i.3], clause 7.5.2.1.1					
TSS reference	ISUP-SIP-ISUP/SS/MCID								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Ensure that the SUT rejects a l	MCID request - MCID not supported by the OLE I-MGCF Ensure that the SUT rejects a MCID request by sending an IRS with the MCID response indicator set to "MCID not included". SIP-I to ISUP interworking.							
SIP parameter	183 Session Progress: Conten								
values	body INFO: Content-Type: application	n/ISUF	P: IRS encapsu	lated in	the MIME body				
ISUP parameter values			,						
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	183 Session Progress(IDR)	+		+	IDR				
	INFO(IRS)	<b>→</b>		<b>→</b>	IRS				
	200 OK INFO	<b>←</b>							
	180 Ringing(ACM) ← ← ACM								
	200 OK INVITE(ANM)	<b>←</b>		<b>←</b>	ANM				
			Conversation						
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	<b>←</b>		+	RLC				

TP407006	SIP reference: RFC 32	61 [4]		Q.731	ISUP reference: Q.1912.5 [1], .7 [i.3], clause 7.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/MCID								
SIP selection criteria									
ISUP selection criteria	PICS 1/7								
Test purpose	<ul> <li>MCID information passed and set correctly - outgoing         Ensure that a received IDR is transferred transparently into the national network, the subsequent IRS being transferred into the international network so that the country code in the address signals of the calling party number is added and the nature of address indicator is set to "international number":         <ul> <li>the IDR request is transferred into the national network;</li> <li>The IRS is received from the national network having the calling party number coded as an "international number". Calling party sub-address in ATP.</li> </ul> </li> </ul>								
SIP parameter values	183 Session Progress: Content body INFO: Content-Type: applicatio	t-Type	: applicatio	n/ISUP; IDF	R encapsulated in the MIME				
ISUP parameter values	71		,	-1	,				
Comments	SIP-I		SU	Γ	ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	183 Session Progress(IDR)	+		<b>+</b>	IDR				
	INFO(IRS)	<b>→</b>		→	IRS				
	200 OK INFO	+							
	180 Ringing(ACM) ← ← ACM								
	200 OK INVITE(ANM)	+		+	ANM				
			Convers	ation					
	BYE(REL)	<b>→</b>		→	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP407007	SIP reference: RFC	3261	[4]		ISUP reference:			
					Q.1912.5 [1],			
				Q.7	731.7 [i.3], clause 7.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/MCID							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Successful MCID request with calling sub-address O-MGCF Ensure that the SUT can successfully reply to an 183 Session Progress (IDR) having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included", the calling party number and a calling sub-address in the access transport parameter. ISUP to SIP-I interworking.							
SIP parameter values	183 Session Progress: Conbody INFO: Content-Type: applic	•			IDR encapsulated in the MIME d in the MIME body			
ISUP parameter values				•				
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	IDR	+		+	183 Session Progress(IDR)			
	IRS	<b>→</b>		→	INFO(IRS)			
				+	- 200 OK INFO			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversa	tion				
	REL	<b>→</b>		7	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP407008	SIP reference: RFC 3261 [4]				-	SUP reference: Q.1912.5 [1], 7 [i.3], clause 7.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/MCID				4.731.	[1.5], clause 1.5.2.1.1		
SIP selection criteria	1301 -311 -1301 /33/WCID							
ISUP selection criteria								
Test purpose	Successful MCID request with calling sub-address I-MGCF Ensure that the SUT can successfully reply to an IDR having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included", the calling party number and a calling sub-address in the access transport parameter. SIP-I to ISUP interworking.							
SIP parameter values	183 Session Progress: Content body INFO: Content-Type: application	it-Type:	applicat			·		
ISUP parameter values				•		,		
Comments	SIP-I		SI	JT		ISUP		
	INVITE(IAM)	→			<b>→</b>	IAM		
	183 Session Progress(IDR)	+			+	IDR		
	INFO(IRS)	<b>→</b>			<b>→</b>	IRS		
	200 OK INFO	+						
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM)	+			+	ANM		
			Conver	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP407009	SIP reference: RFC	3261	[4]	Q.731	ISUP reference: Q.1912.5 [1], I.7 [i.3], clause 7.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/MCID							
SIP selection criteria								
ISUP selection criteria								
Test purpose	MCID timer (T39) expiry O-MGCF Ensure that call setup is continued (user is alerted) if no IRS is received within timer T39 expiry, after having sent the IDR with MCID request indicator set to "MCID requested". ISUP to SIP-I interworking.							
SIP parameter	183 Session Progress: Con	tent-Ty	pe: application/IS	SUP; IDI	R encapsulated in the MIME			
values	body							
ISUP parameter values								
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	IDR	+		+	183 Session Progress(IDR)			
				T39 e	xpiry			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversation					
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP407010	SIP reference: RFC 32	261 [4]		ISUP reference: Q.1912.5 [1], Q.731.7 [i.3], clause 7.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/MCID							
SIP selection criteria								
ISUP selection criteria								
Test purpose	MCID timer (T39) expiry O-MGCF Ensure that call setup is continued (user is alerted) if no IRS is received within timer T39 expiry, after having sent the IDR with MCID request indicator set to "MCID requested". SIP-I to ISUP interworking.							
SIP parameter values	183 Session Progress: Content body INFO: Content-Type: application	,,			•			
ISUP parameter values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	183 Session Progress(IDR)	+		+	IDR			
				Т39 е	expiry			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

# 5.3.8 Call hold (HOLD)

TP408001	SIP reference: RF0	C 3261	[4]		ISUP reference:
					Q.1912.5 [1],
				Q.733 [i.6],	clauses 2.5.2.1.1.1; 2.5.2.1.1.2
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Call hold after answer, requ	uested k	by the origin	nating user	
	messages having the even	t indica	ntor set to "	progress". O	
SIP parameter	INVITE: Content-Type: app	olication	ISUP; CPG	encapsulate	ed in the MIME body
values					
ISUP parameter					
values	IOUR	1			loin i
Comments	ISUP	<u> </u>	SU		SIP-I
	IAM	<b>→</b>		-	11 1 1 1 1 2 (1) (11)
	ACM	+		•	100111191119
	ANM	+		€	200 OK INVITE(ANM)
			Convers		
	CPG(progress, hold)	→		<b>→</b>	
				<b>+</b>	200 OK INVITE
				-	ACK
	CPG(progress, retrieve)	<b>→</b>		)	(0: 0; 00::0::0:/
				+	
				7	ACK
	REL	<b>→</b>		7	BYE(REL)
	RLC	+		<del>(</del>	200 OK BYE(RLC)

TP408002	SIP reference: RF0	3261	[4]		ISUP reference:
				0 722 [; 6] 6	Q.1912.5 [1], lauses 2.5.2.1.1.1; 2.5.2.1.1.2
TSS reference	ISUP-SIP-ISUP/SS/HOLD			Q.733 [1.0], C	iauses 2.5.2.1.1.1, 2.5.2.1.1.2
SIP selection	150P-51P-150P/55/HOLD				
criteria					
ISUP selection criteria					
Test purpose	Call hold after answer, requ	iested k	y the origin	nating user	
	messages having the even	t indica	tor set to "	progress". I-M	
SIP parameter values	INVITE: Content-Type: app	lication	ISUP; CPG	encapsulated	d in the MIME body
ISUP parameter values					
Comments	SIP-I		SU	Τ	ISUP
	INVITE(IAM)	<b>→</b>		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Convers	ation	
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(progress, hold)
	200 OK INVITE	+			
	ACK	<b>→</b>			
	INI\/ITE/CDC against again)	<b>→</b>		<b>→</b>	CDC/programs and risks)
	INVITE(CPG, sendrecv)	+		7	CPG(progress, retrieve)
	200 OK INVITE	<b>→</b>			
	ACK	7			+
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP408003	SIP reference: RF	C 3261	[4]		ISUP reference: Q.1912.5 [1], lauses 2.5.2.1.1.1; 2.5.2.1.1.2			
TSS reference	ISUP-SIP-ISUP/SS/HOLD							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Call hold after answer, requested by the terminating user  Ensure that the notifications that a call is placed on hold and retrieved are sent with CPG messages having the event indicator set to "progress". O-MGCF interworking.							
SIP parameter values	INVITE: Content-Type: app	olication	/ISUP; CPG	encapsulated	I in the MIME body			
ISUP parameter values								
Comments	ISUP		SU	Г	SIP-I			
	IAM	<b>→</b>		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Convers	ation				
	CPG(progress, hold)	+		+	INVITE(CPG, sendonly)			
				<b>→</b>	200 OK INVITE			
				<b>←</b>	ACK			
	CPG(progress, retrieve)	+		<b>←</b>	INVITE(CPG, sendrecv)			
				→	200 OK INVITE			
				<b>←</b>	ACK			
	REL	→		→	BYE(REL)			
	RLC	<b>←</b>		←	200 OK BYE(RLC)			

TP408004	SIP reference: RF0	3261	[4]		ISUP reference: Q.1912.5 [1],
				Q.733 [i.6]. c	lauses 2.5.2.1.1.1; 2.5.2.1.1.2
TSS reference	ISUP-SIP-ISUP/SS/HOLD				,
SIP selection					
criteria					
ISUP selection criteria					
Test purpose	Call hold after answer, requ	uested k	y the termi	nating user	
	messages having the even	t indica	tor set to "	progress". I-M	
SIP parameter values	INVITE: Content-Type: app	lication	ISUP; CPG	encapsulated	I in the MIME body
ISUP parameter values					
Comments	SIP-I		SU	Γ	ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		<b>+</b>	ANM
			Convers		
	INVITE(CPG, sendonly)	+		<b>←</b>	CPG(progress, hold)
	200 OK INVITE	→			
	ACK	+			
	INVITE(CPG, sendrecv)	+		+	CPG(progress, retrieve)
	200 OK INVITE	<b>→</b>			
	ACK	+			
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	<b>←</b>		+	RLC

TP408005	SIP reference: RF	C 3261	[4]		ISUP reference: Q.1912.5 [1], , clauses 2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2
TSS reference	ISUP-SIP-ISUP/SS/HOLD				
SIP selection criteria					
ISUP selection criteria	PICS 8/1				
Test purpose	Call hold after alerting, re Ensure that when an outgo notifications are sent with (	oing call	is placed or	n hold and retr	ieved after alerting the rking.
SIP parameter values	INVITE: Content-Type: app				
ISUP parameter values					
Comments	ISUP		SU <sup>-</sup>	Т	SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	CPG(progress, hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly) 200 OK INVITE
				<b>→</b>	ACK
	CPG(progress, retrieve)	<b>→</b>		<b>→</b>	INVITE(CPG, sendrecv)
				+	200 OK INVITE
				<b>→</b>	ACK
	ANM	+		+	200 OK INVITE(ANM)
			Convers	ation	
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP408006	SIP reference: RF	C 3261	[4]		ISUP reference: Q.1912.5 [1], , clauses 2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2		
TSS reference	ISUP-SIP-ISUP/SS/HOLD						
SIP selection criteria							
ISUP selection criteria	PICS 8/1						
Test purpose	Call hold after alerting, requested by the calling user Ensure that when an outgoing call is placed on hold and retrieved after alerting the notifications are sent with CPG messages. I-MGCF interworking.						
SIP parameter values	INVITE: Content-Type: app	olication	/ISUP; CPG	encapsulated	I in the MIME body		
ISUP parameter values							
Comments	SIP-I		SU <sup>-</sup>	Γ	ISUP		
	INVITE(IAM)	<b>→</b>		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(progress, hold)		
	200 OK INVITE	+					
	ACK	<b>→</b>					
	INVITE(CPG, sendrecv)	<b>→</b>		<b>→</b>	CPG(progress, retrieve)		
	200 OK INVITE	+					
	ACK	<b>→</b>					
	200 OK INVITE(ANM)	+		+	ANM		
	,		Convers	ation			
	BYE(REL)	<b>→</b>		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP408007	SIP reference: R	FC 3261 [4	1]	l	ISUP reference:
					Q.1912.5 [1],
				Q.7	64 [i.12], clause 2.3
TSS reference	ISUP-SIP-ISUP/SS/HOLI	D			
SIP selection					
criteria					
ISUP selection criteria					
Test purpose	Call hold after answer,	release of	the call by t	he calling s	erved user
	service. O-MGCF interwo	orking.		•	er who activated the Call hold
SIP parameter	INVITE: Content-Type: a	pplication/I	SUP; CPG e	ncapsulated	in the MIME body
values				•	-
ISUP parameter					
values					
Comments	ISUP		SUT		SIP-I
	IAM	→		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversati	on	
	CPG(progress, hold)	<b>→</b>		→	INVITE(CPG, sendonly)
				+	200 OK INVITE
				<b>→</b>	ACK
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP408008	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.764 [i.12], clause 2.3				
TSS reference	ISUP-SIP-ISUP/SS/HOLD					,			
SIP selection criteria									
ISUP selection criteria									
Test purpose	Call hold after answer, re Ensure that a call in the he service. I-MGCF interworki	ld state				erved user er who activated the Call hold			
SIP parameter values	INVITE: Content-Type: app		/ISUP; CPG	encap	sulated	in the MIME body			
ISUP parameter values									
Comments	SIP-I		SUT			ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
			Convers	ation					
	INVITE(CPG, sendonly)	<b>→</b>			<b>→</b>	CPG(progress, hold)			
	200 OK INVITE	+							
	ACK	<b>→</b>							
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			+	RLC			

TP408009	SIP reference: R	FC 3261 [4]			ISUP reference: Q.1912.5 [1], 64 [i.12], clause 2.3
TSS reference	ISUP-SIP-ISUP/SS/HOL	D			
SIP selection criteria					
ISUP selection criteria					
Test purpose	Call hold after answer, Ensure that a call in the h Call hold service. O-MG0	neld state ca	n be released l		ting user ser who did not activate the
SIP parameter	INVITE: Content-Type: a			psulated	d in the MIME body
values	,		•	•	ŕ
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	→		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversation	•	· ·
	CPG(progress, hold)	+		+	INVITE(CPG, sendonly)
				<b>→</b>	200 OK INVITE
				+	ACK
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		<del>-</del>	200 OK BYE(RLC)

TP408010	SIP reference: RF	SIP reference: RFC 3261 [4]				SUP reference: Q.1912.5 [1], 64 [i.12], clause 2.3
TSS reference	ISUP-SIP-ISUP/SS/HOLD		- 1			
SIP selection criteria						
ISUP selection criteria						
Test purpose	Call hold after answer, re Ensure that a call in the he Call hold service. I-MGCF	ld state	can be relea			ing user er who did not activate the
SIP parameter values	INVITE: Content-Type: app			encaps	sulated	in the MIME body
ISUP parameter values						
Comments	SIP-I		SUT			ISUP
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM
	180 Ringing(ACM)	+			+	ACM
	200 OK INVITE(ANM)	+			+	ANM
			Conversa	ation		
	INVITE(CPG, sendonly)	+			+	CPG(progress, hold)
	200 OK INVITE	<b>→</b>				
	ACK	+				
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	+			+	RLC

TP408011	SIP reference: R	FC 3261 [4]	1		ISUP reference: Q.1912.5 [1], 64 [i.12], clause 2.3
TSS reference	ISUP-SIP-ISUP/SS/HOLI	D			
SIP selection criteria					
ISUP selection criteria					
Test purpose	Call hold after alerting, Ensure that a held call ca without retrieving the call	ın be releas	ed by the user		user vated the Call hold service
SIP parameter values	INVITE: Content-Type: a	pplication/IS	SUP; CPG enca	psulated	in the MIME body
ISUP parameter values					
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
			Ringing		
	CPG(progress, hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)
				+	200 OK INVITE
				<b>→</b>	ACK
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		<b>+</b>	200 OK BYE(RLC)

TP408012	SIP reference: RF	C 3261 [4	]		ISUP reference: Q.1912.5 [1], 64 [i.12], clause 2.3
TSS reference	ISUP-SIP-ISUP/SS/HOLD		l l		o : [2], o.aaoo 2.0
SIP selection criteria					
ISUP selection criteria					
Test purpose	Call hold after alerting, re Ensure that a held call can without retrieving the call. I	be releas	ed by the use		user vated the Call hold service
SIP parameter values	INVITE: Content-Type: app	olication/IS	SUP; CPG en	capsulated	in the MIME body
ISUP parameter values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		+	ACM
			Ringing		
	INVITE(CPG, sendonly)	<b>→</b>		→	CPG(progress, hold)
	200 OK INVITE	+			
	ACK	<b>→</b>			
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

# 5.3.9 Call Waiting (CW)

TP409001	SIP reference: RFC	3261	[4]	ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1				
TSS reference:	ISUP-SIP-ISUP/SS/CW							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Call waiting indication in AC  Ensure that a call can be su waiting call. O-MGCF interv	ıccessf	•	the <b>ACN</b>	I indicates that it this call a			
SIP parameter values	180 Ringing: Content-Type:			encapsu	lated in the MIME body			
ISUP parameter values	ACM: Generic notification in	ndicato	"Call is a waiting	call"				
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		→	INVITE(IAM)			
	ACM(waiting)	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversation	•				
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP409002	SIP reference: RF	C 3261	[4]	-	ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/CW		<u>.</u>						
SIP selection criteria									
ISUP selection criteria									
Test purpose	Call waiting indication in A  Ensure that a call can be s waiting call. I-MGCF interw	uccessf	ully established	if the <b>ACM</b>	I indicates that this call is a				
SIP parameter values	180 Ringing: Content-Type	e: applica	ation/ISUP; ACN	1 encapsul	ated in the MIME body				
ISUP parameter values	ACM: Generic notification	indicator	"Call is a waitin	g call"					
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM(waiting)				
	200 OK INVITE(ANM) ← ANM								
			Conversation	1					
	BYE(REL)	→		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP409003	SIP reference	e: RFC 3261	Q.7:	ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/C	CW						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Call waiting indication  Ensure that a call car waiting call. O-MGCF	n be successf		if the CF	<b>PG</b> indicates that this call is a			
SIP parameter values				G encap	sulated in the MIME body			
ISUP parameter values	CPG: Generic notifica	ation indicator	"Call is a waitir	ng call"				
Comments	ISUP		SUT		SIP-I			
	IAM	→		<b>→</b>	INVITE(IAM)			
	ACM	+		+	183 Session Progress(ACM)			
	CPG(waiting)	+		+	180 Ringing(CPG)			
	ANM ← 200 OK INVITE(ANM)							
			Conversation	1				
	REL	→		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP409004	SIP reference: RFC 32	261 [4]	'	ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/CW							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Call waiting indication in CPG		·		·			
SIP parameter values ISUP parameter values	Ensure that a call can be succe waiting call. I-MGCF interworki 180 Ringing: Content-Type: ap  CPG: Generic notification indic	ng. plication	on/ISUP; CPG end	capsul				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	183 Session Progress ACM)	+		+	ACM			
	180 Ringing(CPG)	+		+	CPG(waiting)			
	200 OK INVITE(ANM)	+		+	ANM			
	,		Conversation	•				
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP409005	SIP reference: F	RFC 32	61 [4]	ISUP reference: Q.1912.5 [1], Q.733 [i.6], clause 1.5.2.1.1	
TSS reference	ISUP-SIP-ISUP/SS/CW				
SIP selection					
criteria					
ISUP selection					
criteria	Harmanianta tha societa	11			
Test purpose	User rejects the waiting Ensure that the SUT past the waiting call. O-MGCI	ss on a		#21 (d	call rejected) if a busy user rejects
SIP parameter	180 Ringing: Content-Ty	/pe: ap	plication/ISUP;	ACM o	r CPG encapsulated in the MIME
values	body				•
	480 Temporarily unavail	able: C	ontent-Type: ap	plication	on/ISUP; REL encapsulated in the
	MIME body				
ISUP parameter	ACM or CPG: Generic n	otificati	ion indicator "Ca	ll is a v	vaiting call"
values	REL: Cause #21 (call rej	jected)			
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM(waiting)	+		+	180 Ringing(ACM)
	REL(#21)	+		+	480 Temporarily Unavailable(REL)
	RLC	<b>→</b>		<b>→</b>	ACK

TP409006	SIP reference: RFC 3261 [4] ISUP reference:								
				Q.1912.5 [1],					
			Q.73	3 [i.6	], clause 1.5.2.1.1				
TSS reference	ISUP-SIP-ISUP/SS/CW								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	User rejects the waiting call								
	Ensure that the SUT pass on a REL w	ith ca	use #21 (call rej	ected	d) if a busy user rejects				
	the waiting call. I-MGCF interworking.								
SIP parameter	180 Ringing: Content-Type: application	า/ISU	P; ACM or CPG	enc	apsulated in the MIME				
values	body								
	480 Temporarily unavailable: Content-	Type	application/ISU	IP ; R	EL encapsulated in the				
	MIME body								
ISUP parameter	ACM or CPG: Generic notification indic	cator	"Call is a waiting	g call'	'				
values	REL: Cause #21 (call rejected)								
Comments	SIP-I SUT ISUP								
	INVITE(IAM)	<b>↑</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM(waiting)				
	480 Temporarily Unavailable(REL)	+		+	REL(#21)				
	ACK	<b>→</b>		<b>→</b>	RLC				

TP409007	SIP reference	: RFC 3261	4]		ISUP reference: Q.1912.5 [1], [i.6], clause 1.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/C	W					
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Call waiting ignored (expiry of call waiting supervision timer)						
					wer from user, user alerted) if		
	a busy user does not						
SIP parameter		·Type: applica	ition/ISUP ; ACM	l or CPG	encapsulated in the MIME		
values	body						
		ailable: Cont	ent-Type: applica	ition/ISUF	P; REL encapsulated in the		
	MIME body						
ISUP parameter	ACM or CPG: Generic				call"		
values	REL: Cause #19 (no a	answer from u		l)	T=== -		
Comments	ISUP		SUT		SIP-I		
	IAM	→		→	INVITE(IAM)		
	ACM(waiting)	+		+	180 Ringing(ACM)		
	T9 expiry						
	CASE A						
	REL(#19)	→		<b>→</b>	BYE(REL)		
	RLC	+		<b>←</b>	200 OK BYE(RLC)		
				+	487 Request Terminated		
				→	ACK		
	CASE B						
	REL(#19)	→		→	CANCEL		
	RLC	+		<b>←</b>	200 OK CANCEL		
				<b>←</b>	487 Request Terminated		
				<b>→</b>	ACK		

TP409008	SIP reference: RFC	3261	[4]		-	SUP reference: Q.1912.5 [1],		
					2.733	[i.6], clause 1.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/CW							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Call waiting ignored (expi	Call waiting ignored (expiry of call waiting supervision timer)						
	Ensure that the SUT pass of	n a RE	L with cause	e #19 (no	answ	er from user, user alerted) if		
	a busy user does not answe	er the v	vaiting call. I	-MGCF ii	nterwo	orking.		
SIP parameter	180 Ringing: Content-Type:							
values	body					•		
	480 Temporarily unavailable	e: Cont	ent-Type: ap	plication	/ISUP	; REL encapsulated in the		
	MIME body			•		•		
ISUP parameter	ACM or CPG: Generic notifi	ication	indicator "Ca	all is a wa	aiting	call"		
values	REL: Cause #19 (no answe	r from	user, user al	erted)	Ū			
Comments	SIP-I		SUT			ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	180 Ringing(ACM)	+			+	ACM(waiting)		
	T9 expiry							
	BYE(REL)	<b>→</b>			<b>→</b>	REL(#19)		
	200 OK BYE(RLC)	+			+	RLC		
	487 Request Terminated	+						
	ACK	<b>→</b>						

# 5.3.10 Call Diversion (CFB, CFNR, CFU, CD)

TP410001	SIP reference: RF0	C 3261	[4]			ISUP reference:		
				Q.1	912.5	[1], Q.732 [i.4], clause 2.5		
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	"Call is diverting" indication	receive	ed in 180 Ri	inging				
	Verify that a call can be such	ccessful	lly establish	ed, if div	versior	n occurs. The <b>ACM</b> contains		
	the generic notification in	dicator				the call diversion information		
	and the redirection number							
	The Redirection reason is s							
	CPG (alerting) is coded as	if it has	been mapp	ed from	the 1	80 Ringing (CPG).		
	O-MCGF interworking.							
SIP parameter		ntent-Ty	pe: applicat	tion/ISU	P; AC	M encapsulated in the MIME		
values	body							
	180 Ringing: Content-Type					ulated in the MIME body		
ISUP parameter	ACM: BCI Called party sta	ıtus indi	cator "No in	dication	ו"			
values	Generic notification							
	Call diversion inform							
	Redirection number							
Comments	CPG: Event indicator=alert	ing	SUT			SIP-I		
Comments		<b>→</b>	501		<b>→</b>			
	IAM	<del>  7</del>			_	INVITE(IAM)		
	ACM(no indication)				<del>(</del>	183 Session Progress(ACM)		
	CPG(alerting)	+			<u>+</u>	180 Ringing(CPG)		
	ANM	+			+	200 OK INVITE(ANM)		
			Conversa	ation				
	REL	<b>→</b>			<b>→</b>	BYE(REL)		
	RLC	<b>←</b>			<b>←</b>	200 OK BYE(RLC)		

TP410002	SIP reference: RFC 32	61 [4]		0.40	-	SUP reference:			
<b></b>	INCLES OF TOTAL PROPERTY.			Q.19	12.5 <u>[</u>	1], Q.732 [i.4], clause 2.5			
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion								
SIP selection criteria									
ISUP selection criteria									
Test purpose	"Call is diverting" indication received in CPG								
	Verify that a call can be successfully established, if diversion occurs. The <b>ACM</b> contains the <b>generic notification indicator</b> set to "call is diverting", the <b>call diversion information</b> and the <b>redirection number.</b> The Redirection reason is set to <b>CV_redirection_reason</b> . 180 Ringing (CPG (alerting)) is coded as if it has been mapped from the CPG. I-MCGF interworking.								
SIP parameter	183 Session Progress: Content	t-Type:	applicat	tion/ISUP	; ACN	encapsulated in the MIME			
values	body	71	• •		•	•			
	180 Ringing: Content-Type: ap	plicatio	n/ISUP:	CPG end	apsul	ated in the MIME body			
ISUP parameter		ACM: BCI Called party status indicator "No indication"							
values	Generic notification								
	Call diversion information	n							
	Redirection number								
	CPG: Event indicator=alerting								
Comments	SIP-I		S	UT		ISUP			
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM			
	183 Session Progress(ACM)	+			+	ACM(no indication)			
	180 Ringing(CPG)	+			+	CPG(alerting)			
	200 OK INVITE(ANM)	+			+	ANM			
	Conversation								
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)	+			+	RLC			

CV_redirection_reason, TP410001, TP410002						
VA_1	User busy					
VA_2	Unconditional					
VA 3	Deflection immediate response					

TP410003	SIP reference: RFC	3261	[4]	O 1012 F [	ISUP reference:		
TSS reference	ISUP-SIP-ISUP/SS/Call Div	oroion		Q.1912.5 [	1], Q.732 [i.4], clause 2.5.2.1.1		
SIP selection	150P-51P-150P/55/Call DIV	ersion					
criteria							
ISUP selection							
criteria							
Test purpose  SIP parameter	"Call diversion may occur" received in 180 Ringing(ACM)  Verify that a call can be successfully established, if diversion may occur. The encapsulated ACM indicates that "call diversion may occur" in the optional backward call indicators. The following CPG contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number, if diversion occurs. The CPG (progress) contains CV_redirection_reason in call diversion information and also Redirection number. The CPG (alerting) is coded as if it has been mapped from ACM, with RnNbRes parameter (optional).  O-MCGF interworking.  180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body						
values	183 Session Progress: Content-Type: application/ISUP; CPG encapsulated in the MIME body						
ISUP parameter values	ACM: BCI Called party status indicator "subscriber free", Optional backward call indicator:  "Call diversion may occur"  CPG: Event information=progress, Call diversion information; Generic notification;  Redirection number  CPG: Event information=alerting						
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM(free)	+		<b>←</b>	180 Ringing(ACM)		
	CPG	+		+	183 Session Progress(CPG)		
	CPG(alerting)	+		+	183 Session Progress(CPG)		
	ANM	+		+	200 OK INVITE(ANM)		
			Conversa	ition			
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP410004	SIP reference: RFC 320	61 [4]	Q.19	_	SUP reference: Q.732 [i.4], clause 2.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	"Call diversion may occur" received in ACM  Verify that a call can be successfully established, if diversion may occur. The ACM indicates that "call diversion may occur" in the optional backward call indicators. The following CPG contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number, if diversion occurs.  The CPG (progress) contains CV_redirection_reason in call diversion information and also Redirection number. The CPG (alerting) is coded as if it has been mapped from ACM, with RnNbRes parameter (optional).  I-MCGF interworking.							
SIP parameter	180 Ringing: Content-Type: app	olication/	ISUP; ACM	encapsu	lated in the MIME body			
values	183 Session Progress: Content							
	body		•		•			
ISUP parameter values	"Call diversion may occu CPG: Event information=progre Redirection number CPG: Event information=alerting	ACM: BCI Called party status indicator "subscriber free", Optional backward call indicator:  "Call diversion may occur"  CPG: Event information=progress, Call diversion information; Generic notification;  Redirection number						
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		→	IAM			
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ACM(free)			
	183 Session Progress(CPG)	<del>(</del>		+	CPG			
	183 Session Progress(CPG)	<b>←</b>		←	CPG(alerting)			
	200 OK INVITE(ANM)	<b>←</b>		+	ANM			
			onversation					
	BYE(REL)	<b>→</b>		→	REL			
	200 OK BYE(RLC)	<b>←</b>		+	RLC			

CV_redirection_reason TP410003, TP410004						
VA_1	No reply					
VA_2	Deflection during alerting					

TP410005	SIP reference: RF	C 3261	[4]	0.4040.5	ISUP reference:		
TCC votovovos				Q.1912.5	[1], Q.732 [i.4], clause 2.4.2		
TSS reference	ISUP-SIP-ISUP/SS/Call Di	version					
SIP selection criteria							
ISUP selection							
criteria							
Test purpose	Multiple diversions -Verify that a call can be successfully established, if multiple diversion occur  Several messages each containing the call diversion information are received, as if multiple forwardings have occurred.  The CV redirection reason is used as redirection reason.						
	The Redirection number re						
	O-MCGF interworking.						
SIP parameter		ntent-Ty	pe: applicat	tion/ISUP; AC	CM encapsulated in the MIME		
values	body	·		ŕ	•		
	183 Session Progress: Cor	ntent-Ty	pe: applicat	tion/ISUP; CF	PG encapsulated in the MIME		
	body						
	180 Ringing: Content-Type				sulated in the MIME body		
ISUP parameter values	ACM: BCI Called party status indicator "No indication" Generic notification Call diversion information Redirection reason unconditional Redirection number CPG1: Event information=progress Generic notification Call diversion information Redirection reason CV_redirection_reason Redirection number Redirection number restriction						
_	CPG2: Event information=	alerting,		number res			
Comments	ISUP	<u> </u>	SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM(no indication)	+		+	183 Session Progress(ACM)		
	CPG1	+		<del>-</del>	183 Session Progress(CPG)		
	CPG2(alerting)	+		+	180 Ringing(CPG)		
	ANM	+		←	200 OK INVITE(ANM)		
			Conversa				
	REL	→		→	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

TP410006	SIP reference: RFC 32	61 [4]			ISUP reference:						
				Q.1912.5 [	1], Q.732 [i.4], clause 2.4.2						
TSS reference	ISUP-SIP-ISUP/SS/Call Divers	ISUP-SIP-ISUP/SS/Call Diversion									
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	Multiple diversions -Verify that a call can be successfully established, if multiple										
		diversion occur									
	Several messages each contain		e <b>call di</b>	version infor	mation are received, as if						
	multiple forwardings have occu										
	The CV_redirection_reason is										
	The Redirection number restric	tion pa	rameter	is passed on.							
OID	I-MCGF interworking.	_		· //OLID A O	A L L L A BAIRAT						
SIP parameter	183 Session Progress: Content	-Type:	applicat	tion/ISUP; ACI	vi encapsulated in the MIME						
values	body	T	!:4	ian /ICLID: CD/	Concernsulated in the MINAT						
	183 Session Progress: Content	- rype:	applicat	ion/iSUP; CP	s encapsulated in the MilME						
	body 180 Ringing: Content-Type: ap	alicatio	n/ICI ID:	CPG onconsu	ulated in the MIME body						
ISUP parameter	ACM: BCI Called party status				ilated in the Milvic body						
values	Generic notification	iiiuicaii	וו טאו וכ	luication							
Values	Call diversion information	n Redi	rection r	eason uncond	litional						
	Redirection number	ii ixcai	i Colloii i	cason anconc	intorial						
	CPG: Event information=progre	ess									
	Generic notification	,00									
	Call diversion information	n Redi	rection r	eason CV red	direction reason						
	Redirection number			_	_						
	Redirection number rest	riction									
	CPG: Event information=alertin	g, Red	irection	number restric	ction						
Comments	SIP-I			UT	ISUP						
	INVITE(IAM)	<b>→</b>		→	IAM						
	183 Session Progress(ACM)	+		+	ACM(no indication)						
	183 Session Progress(CPG)	+		+	CPG1						
	180 Ringing(CPG)	+		+	CPG2(alerting)						
	200 OK INVITE(ANM)	+		+	ANM						
			Conve	rsation							
	BYE(REL)	<b>→</b>		<b>→</b>	REL						
	200 OK BYE(RLC)	+		+	RLC						

CV_redirection_reason, TP410005, TP410006							
VA_1	No reply						
VA_2	Deflection during alerting						
VA_3	User busy						
VA_4	Unconditional						
VA 5	Deflection immediate response						

TP410007	SIP reference: RFC	3261	4]		ISUP reference: , Q.732 [i.4], clause 2.5.2.2.1			
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion		Q.1312.3[1]	, Q.732 [1.4], Clause 2.3.2.2.1			
SIP selection	leer en leer /ee/ean biv	CIGIOII						
criteria								
ISUP selection								
criteria								
Test purpose	Notification procedures for a diverting call - after the diverting exchange							
	Verify that the IUT can succediversion) all the diversion in							
	It has to be checked that the direction:	follow	ing signallir	ng information	is passed on in the forward			
	redirecting num original called n redirection infor	umber	(see note)	;				
	It has to be checked that the	follow	ing signallir	ng information	is passed on in the backward			
	direction:							
		oer res	<b>triction</b> pa	rameter (in AC	CM /CPG /ANM /CON).			
	O-MCGF interworking.							
SIP parameter	INVITE: Content-Type: appl							
values	200 OK INVITE: Content-Ty							
ISUP parameter	IAM: Redirecting number, O			ber, Redirection	on information			
values	ANM: Redirection address re	estriction		_ 1	loin i			
Comments	ISUP		SU		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		<u></u> ←	200 OK INVITE(ANM)			
			Convers					
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			
NOTE: Altered in	Gateways.							

TP410008	SIP reference: RFC	3261	[4]	O 1012		SUP reference:				
TSS reference	Q.1912.5 [1], Q.732 [i.4], clause 2.5.2.2.1  ISUP-SIP-ISUP/SS/Call Diversion									
SIP selection	1001 On 1001/00/04ll Divolololi									
criteria										
ISUP selection										
criteria										
Test purpose	Notification procedures for a diverting call - after the diverting exchange									
	Verify that the IUT can succ diversion) all the diversion in									
	It has to be checked that the direction:									
	redirecting num	ber (se	e note):							
	original called n									
	redirection infor	matior	۱.							
	It has to be checked that the	e follow	ing signallir	ng informa	ation is	s passed on in the backward				
	direction:									
		ber res	striction pa	rameter (	in ACI	M /CPG /ANM /CON).				
	I-MCGF interworking.									
SIP parameter	INVITE: Content-Type: appl									
values	200 OK INVITE: Content-Ty									
ISUP parameter	IAM: Redirecting number, O			per, Redir	rection	information				
values	ANM: Redirection address r	estriction		_		I				
Comments	SIP-I		SUT	Ī		ISUP				
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM				
	180 Ringing(ACM)	+			+	ACM				
	200 OK INVITE(ANM)	+			+	ANM				
	Conversation									
	BYE(REL)	<b>→</b>			<b>→</b>	REL				
	200 OK BYE(RLC)	+			+	RLC				
NOTE: Altered in	Gateways.									

TP410009	SIP reference: RF0	C 3261	[4]	0.73	ISUP reference: Q.1912.5 [1], 1 [i.2], clause 3.5.2.4.1		
TSS reference	ISUP-SIP-ISUP/SS/Call Div	version		Q.13	1 [1.2], Clause 3.3.2.4.1		
SIP selection							
criteria							
ISUP selection criteria	PICS 10/1 AND PICS 1/7						
Test purpose  SIP parameter values	Original called number in the outgoing international gateway Verify that the outgoing international gateway checks and manipulates the original called number according to the procedures as defined for CLIP: Discarding the original called number if case of bilateral agreements. The PTC will send an IAM with OriCdNb. INVITE: Content-Type: application/ISUP; IAM containing an Original called number						
values	encapsulated in the MIME	bouy					
ISUP parameter values	IAM: No original called num	nber pre	esent				
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		→	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	<b>←</b>		<b>←</b>	ANM		
			Conversa	tion			
	BYE(REL)	<b>→</b>		→	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP410010	SIP reference: RFC	3261	[4]		ISUP reference:
					Q.1912.5 [1],
				Q.731	[i.2], clause 4.5.2.1.1
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion			
SIP selection					
criteria					
ISUP selection	PICS 1/7				
criteria					
Test purpose	Original called number in				
					anipulates the original called
	number according to the pr				
	Converting the original				ormat with transparent
	transferral of address pr				
	The PTC will send an IAM w				
SIP parameter				containing an	Original called number called
values	number encapsulated in the				
ISUP parameter	IAM: Original called number	"Interr	national num	nber"	
values					
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	4		+	ACM
	200 OK INVITE(ANM)	4		+	ANM
			Convers	ation	
	BYE(REL)	<b>^</b>		<b>→</b>	REL
	200 OK BYE(RLC)	4		+	RLC

TP410011	SIP reference: RFC	3261 [4]			I	SUP reference:			
					_	Q.1912.5 [1],			
				Q.7	<b>'</b> 31	[i.2], clause 4.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion	·			A 47			
SIP selection									
criteria									
ISUP selection	PICS 1/7								
criteria									
Test purpose	Original called number in	the outgo	oing inte	rnational g	ate	way			
						nipulates the original called			
	number according to the pr								
	Discarding the original								
	The PTC will send an IAM v	with an "ac	ddress no	t available"	Ori	iCdNb.			
SIP parameter				containing a	an (	Original called number called			
values	number encapsulated in the	MIME bo	ody						
ISUP parameter	IAM: No original called num	ber prese	nt						
values									
Comments	SIP-I		SUT	•		ISUP			
	INVITE(IAM)	<b>→</b>		-	<del>}</del>	IAM			
	180 Ringing(ACM)	<b>←</b>		•	<u> </u>	ACM			
	200 OK INVITE(ANM)								
			Conversa	ation					
	BYE(REL)	<b>→</b>		-	<b>&gt;</b>	REL			
	200 OK BYE(RLC)	+		•	F	RLC			

TP410012	SIP reference: RFC	3261	[4]		I	SUP reference:		
						Q.1912.5 [1],		
				Q.	731	[i.2], clause 4.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion						
SIP selection								
criteria								
ISUP selection	PICS 1/8							
criteria								
Test purpose	Original called number in							
	Verify that the incoming inte	ernation	nal gateway	checks and	d ma	nipulates the original called		
	number according to the pr	ocedur	es as define	ed for CLIP	. App	plicable tests:		
	Converting the original	called	number to	national fo	rmat	, if necessary (own country		
	code).							
SIP parameter				containing	an C	Original called number called		
values	number encapsulated in the	MIME	body					
ISUP parameter	IAM: Original called numbe	r "Natio	nal number	"				
values								
Comments	SIP-I		SUT	Γ		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	180 Ringing(ACM)	<b>←</b>			<b>←</b>	ACM		
	200 OK INVITE(ANM)    ANM							
			Convers	ation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			<del>(</del>	RLC		

TP410013	SIP reference: R	FC 3261	[4]	G	_	SUP reference: Q.1912.5 [1], [i.2], clause 4.5.2.1.1			
TSS reference	ISUP-SIP-ISUP/SS/Call I	Diversion							
SIP selection									
criteria									
ISUP selection	PICS 10/2 AND PICS 1/7	7							
criteria									
Test purpose	Redirecting number in	the outgo	oing interna	ational ga	itewa	у			
						anipulates the redirecting			
	number according to the								
	Discarding the redire	cting nur	nber if case	of bilate	ral ag	reements.			
SIP parameter	INVITE: Content-Type: a	pplication	/ISUP; IAM	containin	g a R	edirecting number			
values	encapsulated in the MIMI	E body							
ISUP parameter	IAM: No Redirecting num	ber prese	ent						
values									
Comments	SIP-I		SU	Γ		ISUP			
	INVITE(IAM)	→			<b>→</b>	IAM			
	180 Ringing(ACM)	+			+	ACM			
	200 OK INVITE(ANM)	+			+	ANM			
			Convers	ation					
	BYE(REL)	<b>→</b>			<b>→</b>	REL			
	200 OK BYE(RLC)								

TP410014	SIP reference: RFC	3261	[4]		ı	SUP reference:		
						Q.1912.5 [1],		
				Q.	.731	[i.2], clause 4.5.2.1.1		
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion						
SIP selection								
criteria								
ISUP selection	PICS 1/7							
criteria								
Test purpose	Redirecting number in the	e outgo	oing interna	ational gat	ewa	у		
	Verify that the outgoing inte					nipulates the <b>redirecting</b>		
	number according to the pr							
	Discarding the redirecti							
	The PTC will send an IAM v	with an	"address no	ot available	e" Rg	Nb.		
SIP parameter	INVITE: Content-Type: app	lication	/ISUP; IAM	containing	a R	edirecting number		
values	encapsulated in the MIME b	oody				-		
ISUP parameter	IAM: No Redirecting number	er prese	ent					
values								
Comments	SIP-I		SU	Γ		ISUP		
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM		
	180 Ringing(ACM)	+			<b>←</b>	ACM		
	200 OK INVITE(ANM) ← ← ANM							
	,		Convers	ation				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			<b>←</b>	RLC		

TP410015	SIP reference: RFC	3261	[4]		Q.732	SUP reference: Q.1912.5 [1], [i.4], clause 2.5.2.3, [i.2], clause 3.5.2.3	
TSS reference:	ISUP-SIP-ISUP/SS/Call Div	ersion					
SIP selection criteria							
ISUP selection criteria	PICS 1/7						
Test purpose	Redirecting number in the outgoing international gateway  Verify that the outgoing international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP:  Converting the redirecting number to international format with transparent transferral of address presentation restriction indicator.  The PTC will send an IAM with a national significant RgNb.						
SIP parameter values	INVITE: Content-Type: appl number" encapsulated in the			containin	ıg a Re	edirecting number "National	
ISUP parameter values	IAM: Redirecting number "Ir	nternati	onal numbe	er"			
Comments	SIP-I		SU			ISUP	
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM	
	180 Ringing(ACM)	+			+	ACM	
	200 OK INVITE(ANM)	+			+	ANM	
			Convers	ation			
	BYE(REL)	<b>→</b>			<b>→</b>	REL	
	200 OK BYE(RLC)	+			+	RLC	

TP410016	SIP reference: RFC	3261	[4]		ISUP reference: Q.1912.5 [1], 32 [i.4], clause 2.5.2.3, 31 [i.2], clause 3.5.2.3
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion			
SIP selection criteria					
ISUP selection criteria	PICS 1/8				
Test purpose	number according to the pi	ernation rocedur ing nur	al gateway of es as define mber to nation	checks and r d for CLIP:	nanipulates the <b>redirecting</b> if necessary (own country
SIP parameter values	INVITE: Content-Type: app "International number" enca				Redirecting number
ISUP parameter values	IAM: Redirecting number "r			iL body	
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		<b>←</b>	ANM
			Conversa	tion	
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

TP410017	SIP reference: RF	C 3261	[4]	Q.732	SUP reference: Q.1912.5 [1], 2 [i.4], clause 2.5.2.3, 1 [i.2], clause 3.5.2.3				
TSS reference	ISUP-SIP-ISUP/SS/Call Di	iversion	1						
SIP selection criteria									
ISUP selection criteria	PICS 1/8 AND 10/4								
Test purpose	Redirecting number in the incoming international gateway Verify that the incoming international gateway checks and manipulates the redirecting number according to the procedures as defined for CLIP: Adding a prefix to an international redirecting number. The PTC will send an IAM with RgNb.								
SIP parameter values	INVITE: Content-Type: app encapsulated in the MIME		/ISUP; IAM cor	taining a R	edirecting number				
ISUP parameter values	IAM: Redirecting number								
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	← ← ANM							
			Conversation	n					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP410018	SIP reference: RF	C 3261	[4]		ISUP reference: Q.1912.5 [1], 732 [i.4], clause 2.5.2.4, 731 [i.2], clause 3.5.2.4				
TSS reference	ISUP-SIP-ISUP/SS/Call Di	version							
SIP selection criteria									
ISUP selection criteria	PICS 10/5 AND PICS 1/8								
Test purpose	Redirection number in the incoming international gateway  Verify that the incoming international gateway checks and manipulates the redirection number according to the procedures defined for COLP:  discarding the redirection number in case of bilateral agreements; removes the redirection number restriction parameter.								
SIP parameter values	number encapsulated in the	e MIME ype: ap	body plication/ISI	JP; ANM co	CM containing a Redirection ntaining a Redirection address				
ISUP parameter values	ACM: Called party status=r Generic notification Call diversion inform No Redirection num ANM: No Redirection numb	no indica nation R nber	ation Redirection r	eason unco	nditional				
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM(no indication)	+		+	183 Session Progress(ACM)				
	CPG								
	ANM	← 200 OK INVÎTE(ANM)							
			Conversa	ition	, , ,				
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP410019	SIP reference: RFC	C 3261 [4]			ISUP reference: Q.1912.5 [1], .732 [i.4], clause 2.5.2.3, .731 [i.2], clause 3.5.2.3			
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion						
SIP selection criteria								
ISUP selection criteria	PICS 1/7							
Test purpose	Redirection number in the outgoing international gateway  Verify that the outgoing international gateway checks and manipulates the redirection  number according to the procedures defined for COLP:  Converting the redirection number to national format, if necessary (own country code):  1. the PTC will provide the necessary stimulus;  2. ACM with CDInf, GenNot = "call is diverting" and an international RnNb with own CC.							
SIP parameter	183 Session Progress: Con	tent-Ty	pe: application	on/ISUP;	ACM containing a Redirection			
values	number "International numb			the MIM	E body			
ISUP parameter	ACM: Called party status=n	o indic	ation					
values	Generic notification							
	Call diversion inform			ason und	conditional			
	Redirection number	"Natior			la:a ·			
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		-				
	ACM(no indication)	<del>(</del>		•	100 0000:01: 10g:000(:1011)			
	CPG	+		•	133 1 9 9			
	ANM	+			200 OK INVITE(ANM)			
			Conversat					
	REL	<b>→</b>			_ : _ (: : _ /			
	RLC	+		•	200 OK BYE(RLC)			

TP410020	SIP reference: RFC		[4] ISUP reference: Q.1912.5 [1], Q.732 [i.4], clause 2.5.2.3, Q.731 [i.2], clause 3.5.2.3				
TSS reference	ISUP-SIP-ISUP/SS/Call Div	ersion					
SIP selection criteria							
ISUP selection criteria	PICS 1/8						
Test purpose	Redirection number in the incoming international gateway  Verify that the incoming international gateway checks and manipulates the redirection number according to the procedures defined for COLP:  Converting the redirection number to international format.						
SIP parameter	183 Session Progress: Con	tent-Ty	pe: applicat	ion/ISUF	P; ACM containing a Redirection		
values	number "National number"						
ISUP parameter	ACM: Called party status=n	o indica	ation		•		
values	Generic notification						
	Call diversion inform	ation R	edirection r	eason u	nconditional		
	Redirection number	"Interna	ational num	ber"			
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>			→ INVITE(IAM)		
	ACM(no indication)	<b>←</b>			← 183 Session Progress(ACM)		
	CPG	+			← 180 Ringing(CPG)		
	ANM						
			Conversa	tion			
	REL	<b>→</b>			→ BYE(REL)		
	RLC	+			← 200 OK BYE(RLC)		

TP410021	SIP reference: RFC	3261	[4]		ISUP reference:				
		Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.3.1							
TSS reference	ISUP-SIP-ISUP/SS/Call Diversion								
SIP selection									
criteria									
ISUP selection	PICS 1/8 AND PICS 10/6								
criteria									
Test purpose	Redirection number in the								
					d manipulates the <b>redirection</b>				
	number according to the pr								
	Adding a prefix to an inte								
					CDInf, GenNot = "call is diverting"				
	and an international RnNb v								
SIP parameter					ACM containing a Redirection				
values	number "International numb			n the MIM	E body				
ISUP parameter	ACM: Called party status=n	o indica	ation						
values	Generic notification								
	Call diversion inform				conditional				
_	Redirection number	Numbe		Κ					
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>			11 * * * * * * * * * * * * * * * * * *				
	ACM(no indication)	+		€	i de deceien i regioco(i tem)				
	CPG ← 180 Ringing(CPG)								
	ANM ← 200 OK INVITE(ANM)								
			Conversa	tion					
	REL	<b>→</b>		7	BYE(REL)				
	RLC	+		+	- 200 OK BYE(RLC)				

## 5.3.11 CONF

TP411001	SIP reference: RFC 3261 [4]	-	Q.1	reference: 912.5 [1], ], clause 1.6.15					
TSS reference	ISUP-SIP-ISUP/SS/CONF								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Generic notification transfer "conference established" and "other party added"  To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:  1. assist a call set up from ISUP to SIP-I;  2. check that the notification "conference established" is received in the CPG from conferee at SIP-I;  3. check the notification "other party added" in the CPG.  O-MGCF interworking.  INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body								
values	in o. content-type. application/1001;	01 0	encapsulated in	uic	WIIWE BOdy				
ISUP parameter values	CPG: Generic notification: conference e CPG: Generic notification: other party a		shed						
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
			Conversation						
	CPG(conference established)	<b>→</b>		<b>→</b>	INFO(CPG)				
				+	200 OK INFO				
	CDC (athor party added)	<b>→</b>		_	INFO(CDC)				
	CPG(other party added)	7		<b>→</b>	INFO(CPG)				
				+	200 OK INFO				
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP411002	SIP reference: R	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15				
TSS reference	ISUP-SIP-ISUP/SS/CONF								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Generic notification transfer "conference established" and "other party added"  To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:  1. Assist a call set up from SIP-I to ISUP;  2. Check that the notification "conference established" is received in the CPG from conferee at the ISUP;  3. Check the notification "other party added" in the CPG.  I-MGCF interworking.								
SIP parameter	INFO: Content-Type: app	olicatio	n/ISUP; CPG e	encaps	sulated in the MIME body				
values			•		·				
ISUP parameter values	CPG: Generic notification CPG: Generic notification			shed					
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversation						
	INFO(CPG)	→		→	CPG(conference established)				
	200 OK INFO								
	INFO(CPG) → CPG(other party added)								
	200 OK INFO	<del>/</del>		+*	Or Storier party added)				
	200 011 1141 0	+							
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP411003	SIP reference: RFC 3261 [4]		Q.1	reference: 912.5 [1], ], clause 1.6.15				
TSS reference	ISUP-SIP-ISUP/SS/CONF							
SIP selection								
criteria								
ISUP selection								
criteria								
SIP parameter values ISUP parameter	Generic notification transfer "conference established" and "isolated"  To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:  1. Assist a call set up from ISUP to SIP-I;  2. Check that the notification "conference established" is received in the CPG from conferee at the SIP-I;  3. Check the notification "isolated" in the CPG.  O-MGCF interworking.  INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body  CPG: Generic notification: conference established							
values	CPG: Generic notification: isolated		OUT		loin i			
Comments	ISUP	_	SUT		SIP-I			
		<del>}</del>		<b>→</b>	INVITE(IAM)			
	1.19.11	<u> </u>		<b>←</b>	180 Ringing(ACM)			
	ANM	<u> </u>		<b>←</b>	200 OK INVITE(ANM)			
			Conversation					
	CPG(conference established)	<b>→</b>		<b>→</b>	INFO(CPG)			
	← 200 OK IN							
	CPG(isolated)	<b>&gt;</b>		<b>→</b>	INFO(CPG)			
				+	200 OK INFO			
	REL -	<del>-</del>		<b>→</b>	BYE(REL)			
		<b>(</b>		<b>←</b>	200 OK BYE(RLC)			

TP411004	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CON	F						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Generic notification transfer "conference established" and "isolated"  To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:  1. assist a call set up from SIP-I to ISUP;  2. check that the notification "conference established" is received in the CPG from conferee at SIP-I;  3. check the notification "isolated" in the CPG.  I-MGCF interworking.							
SIP parameter values	INFO: Content-Type: app	licatio	n/ISUP; CPG	encaps	sulated in the MIME body			
ISUP parameter values	CPG: Generic notification CPG: Generic notification			shed				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation	1				
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference established)			
	200 OK INFO	200 OK INFO ←						
	INFO(CPG) → CPG(isolated)							
	200 OK INFO	+						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	<b>←</b>		+	RLC			

TP411005	SIP reference: RFC 3261 [	ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose	Generic notification transfer "confer				
	To verify that the IUT can successful	ully transfe	er/deliver the r	equire	d notifications in/from the
	CPG message:	015.1			
	Assist a call set up from ISUP to				el in the ODO for an
	2. Check that the notification "conf	erence es	tablished" is re	eceive	a in the CPG from
	conferee at SIP-I; 3. Check the notification "reattache	ad" in the (	CDC		
	O-MGCF interworking.	su iii iiie v	JFG.		
SIP parameter	INFO: Content-Type: application/IS	I IP: CPG	encansulated	in the	MIME body
values	in o. Content-Type. application/10	01,010	ericapsulateu	iii tiic	WIIWE BOdy
ISUP parameter	CPG: Generic notification: conferen	ce establi	shed		
values	CPG: Generic notification: isolated				
	CPG: Generic notification: reattache	ed			
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		→	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversatio	n	
	CPG(conference established)	<b>→</b>		<b>→</b>	INFO(CPG)
				+	200 OK INFO
	CPG(isolated)	<b>→</b>		<b>→</b>	INFO(CPG)
				+	200 OK INFO
	CPG(reattached)	<b>→</b>		<b>→</b>	INFO(CPG)
				+	200 OK INFO
	REL	→	-	→	BYE(REL)
	RLC	+		<b>←</b>	200 OK BYE(RLC)

TP411006	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15		
TSS reference	ISUP-SIP-ISUP/SS/CON	<b>IF</b>					
SIP selection criteria							
ISUP selection criteria							
Test purpose	Generic notification transfer "conference established", "isolated" and "reattached"  To verify that the IUT can successfully transfer/deliver the required notifications in CPG message:						
	<ol> <li>Assist a call set up from SIP-I to ISUP;</li> <li>Check that the notification "conference established" is received in the CPG from conferee at SIP-I;</li> <li>Check the notification "reattached" in the CPG.         <ul> <li>I-MGCF interworking.</li> </ul> </li> </ol>						
SIP parameter	INFO: Content-Type: ap	plication	/ISUP; CPG	encap	sulated in the MIME body		
values							
ISUP parameter values	CPG: Generic notificatio CPG: Generic notificatio CPG: Generic notificatio	n: isolat	ed	shed			
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	→		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
	200 01(1147112(7(1411))		Conversation		AUNI		
	INFO(CPG)	<b>→</b>	Jonversation	<u>'</u>  →	CPG(conference established)		
	200 OK INFO	<del>-</del>			Of O(contenence established)		
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(isolated)		
	200 OK INFO	+					
	INFO(CPG)	<b>&gt;</b>		<b>→</b>	CPG(reattached)		
	200 OK INFO	+					
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	<del>(</del>		+	RLC		

TP411007	SIP reference: RFC 3261 [4	ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CONF				
SIP selection					
criteria					
ISUP selection criteria					
Test purpose	Generic notification transfer "confere disconnected"  To verify that the IUT can successful		·		, ,
	CPG message: 1. assist a call set up from ISUP to 3 2. check the notification "other party O-MGCF interworking.		ected" in the CF	PG.	
SIP parameter values	INFO: Content-Type: application/ISU	P; CPG	encapsulated in	the I	MIME body
ISUP parameter	CPG: Generic notification: conference	e establis	shed		
values	CPG: Generic notification: other part CPG: Generic notification: other part	y added			
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversation	•	
	CPG(conference established)	<b>→</b>		<b>→</b>	INFO(CPG)
				+	200 OK INFO
	CPG(other party added)	<b>→</b>		<b>→</b>	INFO(CPG)
				+	200 OK INFO
	CPG(other party disconnected)	<b>→</b>		<b>→</b>	INFO(CPG)
				+	200 OK INFO
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	<del>_</del>		+	200 OK BYE(RLC)

TP411008	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CON	lF		•				
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	disconnected"				d", "other party added" and "other party			
	To verify that the IUT can successfully transfer/deliver the required notifications in/from t CPG message:  1. Assist a call set up from SIP-I to ISUP;  2. Check the notification "other party disconnected" in the CPG.  I-MGCF interworking.							
SIP parameter	INFO: Content-Type: ap	olication	/ISUP: CPG	encap	sulated in the MIME body			
values								
ISUP parameter	CPG: Generic notificatio	n: confe	rence establ	ished				
values	CPG: Generic notificatio							
0	CPG: Generic notificatio	n: otner		nectea				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		<del>(</del>	ACM			
	200 OK INVITE(ANM)	<b>←</b>		<b>←</b>	ANM			
			Conversatio					
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference established)			
	200 OK INFO	+						
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(other party added)			
	200 OK INFO	+						
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(other party disconnected)			
	200 OK INFO	<del>-</del>			S. Station party disconnected)			
	DVE(DEL)				DEI			
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	<b>←</b>		<b>←</b>	RLC			

TP411009	SIP reference: RFC 3261 [4]	C	JP reference: 2.1912.5 [1], [i.7], clause 1.6.15					
TSS reference	ISUP-SIP-ISUP/SS/CONF							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose  SIP parameter	Generic notification transfer "conference established", and disconnect the conference  To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:  1. Assist a call set up from ISUP to SIP-I;  2. Check that the notification "conference established" is received in the CPG from conferee at ISUP;  3. Release the conference.  O-MGCF interworking.  INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body							
values								
ISUP parameter values	CPG: Generic notification: conference estab	lished						
Comments	ISUP	SUT	SIP-I					
	IAM →		→ INVITE(IAM)					
	ACM ←	,	← 180 Ringing(ACM)					
	ANM +		← 200 OK INVITE(ANM)					
		Conversation						
	CPG(conference established) →		→ INFO(CPG)					
			€ 200 OK INFO					
	REL →	+ +,	→ BYE(REL)					
	RLC ←		€ 200 OK BYE(RLC)					

TP411010	SIP reference: R	FC 326	61 [4]		ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15			
TSS reference	ISUP-SIP-ISUP/SS/CON	F	•		• •			
SIP selection								
criteria								
ISUP selection								
criteria								
SIP parameter values ISUP parameter	Generic notification transfer "conference established", and disconnect the conference  To verify that the IUT can successfully transfer/deliver the required notifications in/from the CPG message:  1. Assist a call set up from SIP-I to ISUP;  2. Check that the notification "conference established" is received in the INFO(CPG) from conferee at SIP-I;  3. Release the conference.  I-MGCF interworking.  INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body  CPG: Generic notification: conference established							
values								
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ACM			
	200 OK INVITE(ANM)	+		<b>←</b>	ANM			
			Conversation					
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference established)			
	200 OK INFO	+						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	<del>-</del>		+	RLC			

## 5.3.12 ECT

TP412001	SIP reference: RFC 3261 [4]		ISUP reference:					
					Q.1912.5 [1],			
			Q.73	2.7 [i	.5], clause 7.5.2.1.1.1 a)			
TSS reference	ISUP-SIP-ISUP/SS/ECT							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Capability of sending a call transfer	num	ber for the	activ	e user			
	Verify that the IUT is able to send the C							
	the service activation parameter "call tr			call tra	ansfer number, received in the			
	ISUP FAC, in an INFO request for the	active	user.					
	O-MGCF interworking.							
SIP parameter	INFO: Content-Type: application/ISUP:	FAC	encapsulat	ed in	the MIME body			
values								
ISUP parameter values	FAC: Generic notification=call transfer	active	e, Call trans	fer nu	ımber (PIXIT)			
Comments	ISUP		SUT		SIP-I			
	IAM	<b>←</b>		<b>→</b>	INVITE(IAM)			
	ACM	1		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
		Conversation						
	FAC(call transfer active, CTNb)	<b>←</b>		<b>→</b>	INFO(FAC)			
				+	200 OK INFO			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP412002	SIP reference: RF	C 3261	ISUP reference: Q.1912.5 [1],			
					Q.734 [i.7], clause 1.6.15	
TSS reference	ISUP-SIP-ISUP/SS/ECT		I		and the first production of th	
SIP selection						
criteria						
ISUP selection criteria						
Test purpose	Capability of sending the	e call tra	nsfer number	for	the active user	
SIP parameter values	the service activation para	meter "c he encap	all transfer" an esulated FAC,	d the	tion parameter "Call transfer active", e call transfer number, received in the SUP FAC for the active user.	
ISUP parameter values	FAC: Generic notification=	call trans	sfer active, Ca	ll trar	nsfer number (PIXIT	
Comments	SIP-I		SUT		ISUP	
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM	
	180 Ringing(ACM)	+		+	ACM	
	200 OK INVITE(ANM)	+		<b>←</b>	ANM	
		•	Conversation			
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active, CTNb)	
	200 OK INFO	+				
	BYE(REL)	→		<b>→</b>	REL	
	200 OK BYE(RLC)	<b>←</b>		<b>←</b>	RLC	

TP412005	SIP reference: RFC 3261 [4]	Q.73	ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)						
TSS reference	ISUP-SIP-ISUP/SS/ECT		*		-				
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Capability of sending the call transfer number for the held user  Verify that the IUT is able to transfer the call transfer number received in an ISUP FAC and sent in INFO request containing the FAC for the held user.  O-MGCF interworking.								
SIP parameter	INFO: Content-Type: application/ISU	P: CPC	encapsulat	ed in	the MIME body				
values	INFO: Content-Type: application/ISU								
ISUP parameter	CPG: Event indicator=progress, Gene	eric no	tification=ho	ld	•				
values	FAC: Generic notification=call transfe	r activ	e, Call transf	er ηι					
Comments	ISUP		SUT		SIP-I				
	IAM	→		<b>→</b>	INVITE(IAM)				
	ACM	<b>←</b>		<b>←</b>	180 Ringing(ACM)				
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)				
			onversatio						
	CPG(hold)	→		<b>→</b>	INVITE(CPG, sendonly)				
				+	200 OK INVITE(recvonly)				
				<b>→</b>	ACK				
	FAC(call transfer active, CTNb)	→		<b>→</b>	INFO(FAC)				
				+	200 OK INFO				
				<b>→</b>	INVITE(sendrecv)				
				+	200 OK INVITE(sendrecv)				
				<b>→</b>	ACK				
	REL	→		<b>→</b>	BYE(REL)				
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)				

TP412006	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15				
TSS reference	ISUP-SIP-ISUP/SS/CONF							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Capability of sending the call transfer number for the active user  Verify that the IUT is able to transfer the call transfer number received in an ISUP FAC and sent in INFO request containing the FAC for the held user.  I-MGCF interworking.							
SIP parameter	INFO: Content-Type: applica	ation/IS	SUP: CPG en	capsu	lated in the MIME body			
values	INFO: Content-Type: applica							
ISUP parameter	CPG: Event indicator=progre							
values	FAC: Generic notification=ca							
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation	1				
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	<b>→</b>						
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active, CTNb)			
	200 OK INFO	<b>←</b>						
	INVITE(sendrecv)	<b>→</b>						
	200 OK INVITE(sendrecv)	+						
	ACK	<b>→</b>						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP412009	SIP reference: RFC 3261 [4]			Į,	SUP reference:				
		Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1							
TSS reference	ISUP-SIP-ISUP/SS/ECT		Q.73	2.7 [I					
SIP selection	1001 -011 -1001 700/201								
criteria									
ISUP selection									
criteria									
Test purpose	Loop prevention procedure - initiat								
	Verify that the SUT is able to transfer								
	INFO request containing the LOP me								
	received in an ISUP LOP in an SIP IN	IFO re	quest contai	ning 1	the ISUP LOP message.				
	O-MGCF interworking.								
SIP parameter	INFO: Content-Type: application/ISUI	P; CPC	encapsulat	ted in	the MIME body				
values	INFO: Content-Type: application/ISUI								
10115	INFO: Content-Type: application/ISUI				the MIME body				
ISUP parameter	CPG: Event indicator=progress, Gene	eric no	tification=ho	Id					
values	LOP: request: Call transfer reference	_							
	LOP: response: Call transfer reference FAC: Generic notification=call transfer active, Call transfer number(PIXIT)								
Comments	ISUP		SUT	lei iiu	SIP-I				
Comments	IAM	<b>→</b>	301	<b>→</b>	INVITE(IAM)				
	ACM	+		<del>-</del>	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
	Conversation								
	CPG(hold)	<b>→</b>	Onversation	·  →	INVITE(CPG, sendonly)				
	Ci C(noid)	<del>'</del>			200 OK INVITE(recvonly)				
				<u>`</u>	ACK				
					KOK				
	LOP(request)	<b>→</b>		<b>→</b>	INFO(LOP)				
	201 (104000)	Ť		<del>-</del>	200 OK INFO				
		+		1					
	LOP(response)	+		+	INFO(LOP)				
		1		<b>→</b>	200 OK INFO				
	FAC(call transfer active, CTNb)	<b>→</b>		<b>→</b>	INFO(FAC)				
				+	200 OK INFO				
				<b>→</b>	INVITE(sendrecv)				
				+	200 OK INVITE(sendrecv)				
				<b>→</b>	ACK				
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC(RLC)	+		<b>←</b>	200 OK BYE				

TP412010	SIP reference: RFC	3261	[4]	ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15						
TSS reference	ISUP-SIP-ISUP/SS/ECT									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	Verify that the SUT is able to containing the ISUP LOP me response received in an ISU LOP message.  I-MGCF interworking.									
SIP parameter	INFO: Content-Type: applica									
values	INFO: Content-Type: applica									
	INFO: Content-Type: applica									
ISUP parameter values	CPG: Event indicator=progre LOP: request: Call transfer r LOP: response: Call transfer FAC: Generic notification=ca	eferen r refere	ce ence		nsfer number(PIXIT)					
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM					
	180 Ringing(ACM)	+		+	ACM					
	200 OK INVITE(ANM)	+		+	ANM					
	Conversation									
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)					
	200 OK INVITE(recvonly)	<b>←</b>								
	ACK	<b>→</b>								
	INFO(LOP)	→		→	LOP(request)					
	200 OK INFO	+								
	INFO(LOP)	+		←	LOP(response)					
	200 OK INFO	<b>→</b>								
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active, CTNb)					
	200 OK INFO	+								
	INIVITE (condragy)	<b>→</b>								
	INVITE(sendrecv)	<del>7</del>								
	200 OK INVITE(sendrecv) ACK	<b>→</b>								
	ACK	7		-						
	BYE(REL)	<b>→</b>			REL					
	200 OK BYE(RLC)	<del>  7</del>		→ ←	RLC					
	ZOU ON BIE(NLO)				INLO					

TP412011	SIP reference: RFC 3261 [4]	Q.73	ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)						
TSS reference	ISUP-SIP-ISUP/SS/ECT								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Loop prevention procedure - unsuccessful on timer expiry  To verify that the SUT is able to transfer the loop request received in an ISUP LOP in an INFO request containing the LOP message. Verify that the connection is unsuccessful if the loop detection procedure is unsuccessful. The connection is released from the remote end.  O-MGCF interworking.								
SIP parameter	INFO: Content-Type: application/ISUP	; CPC	encapsulate	ed in	the MIME body				
values	INFO: Content-Type: application/ISUP	; LOF	encapsulate	ed in	the MIME body				
ISUP parameter	CPG: Event indicator=progress, Gener								
values	LOP: request: Call transfer reference								
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
		С	onversation	)					
	CPG(hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)				
				+	200 OK INVITE(recvonly)				
				<b>→</b>	ACK				
	LOP(request)	<b>→</b>		<b>→</b>	INFO(LOP)				
				<b>←</b>	200 OK INFO				
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		<b>←</b>	200 OK BYE(RLC)				

TP412012	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15				
TSS reference	ISUP-SIP-ISUP/SS/CONF		<b>.</b>		<u> </u>			
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Loop prevention procedure - unsuccessful on timer expiry To verify that SUT is able to transfer the loop request received in an INFO request containing the LOP message in an ISUP LOP message. Verify that the connection is unsuccessful if the loop detection procedure is unsuccessful. The connection is released from the remote end. I-MGCF interworking.							
SIP parameter	INFO: Content-Type: application	ation/IS	SUP; CPG enca	apsu	lated in the MIME body			
values	INFO: Content-Type: application	ation/IS	SUP; LOP enca	psu	lated in the MIME body			
ISUP parameter	CPG: Event indicator=progr							
values	LOP: request: Call transfer	referen	ice					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		<b>←</b>	ACM			
	200 OK INVITE(ANM)	+		<b>←</b>	ANM			
			Conversation					
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	<b>→</b>						
	INFO(LOP)	<b>→</b>		<b>→</b>	LOP(request)			
	200 OK INFO	+						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		<b>←</b>	RLC			

TP412013	SIP reference: RFC 3261 [4]		ISUP reference: Q.1912.5 [1],						
			Q.73	2.7 [	i.5], clause 7.5.2.1.1.1 a)				
TSS reference	ISUP-SIP-ISUP/SS/ECT			-	•				
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Loop prevention procedure - successful on timer expiry  Verify that the SUT is able to transfer the loop request received in an ISUP LOP in an INFO request containing the LOP message. Verify that the connection is successful if the loop detection procedure is unsuccessful.								
SIP parameter	O-MGCF interworking.	D. CDC	` anaonaulat	od in	the MIME hady				
values	INFO: Content-Type: application/ISU INFO: Content-Type: application/ISU	P. FAC	encapsulat	eu in	the MIME body				
values	INFO: Content-Type: application/ISU								
ISUP parameter	CPG: Event indicator=progress, Gen				and Milling Body				
values	LOP: request: Call transfer reference		inodiron–no						
	FAC: Generic notification=call transfer		e. Call transf	er nu	ımber(PIXIT)				
Comments	ISUP		SUT		SIP-I				
	IAM	→		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
	Conversation								
	CPG(hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)				
				+	200 OK INVITE(recvonly)				
				<b>→</b>	ACK				
	LOP(request)	→		<b>→</b>	INFO(LOP)				
				+	200 OK INFO				
	FAC(call transfer active, CTNb)	→		<b>→</b>	INFO(FAC)				
				+	200 OK INFO				
				<b>→</b>	INVITE(sendrecv)				
				+	200 OK INVITE(sendrecv)				
				<b>→</b>	ACK				
	REL	→		<b>→</b>	BYE(REL)				
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)				

TP412014	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1],					
					Q.734 [i.7], clause 1.6.15					
TSS reference	ISUP-SIP-ISUP/SS/ECT									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	Verify that the SUT is able to containing the LOP message	Loop prevention procedure - successful on timer expiry  Verify that the SUT is able to transfer the loop request received in an INFO request containing the LOP message in an ISUP LOP message. Verify that the connection is successful if the loop detection procedure is unsuccessful.  LMGCF interverking.								
SIP parameter	INFO: Content-Type: applica	ation/IS	SUP; CPG e	ncapsu	lated in the MIME body					
values	INFO: Content-Type: applica									
	INFO: Content-Type: applica	ation/IS	SUP; LOP er	ncapsul	ated in the MIME body					
ISUP parameter	CPG: Event indicator=progre			cation=l	nold					
values	LOP: request: Call transfer re	eferen	ce							
	FAC: Generic notification=ca	all tran		Call trar						
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	→		→	IAM					
	180 Ringing(ACM)	+		<b>←</b>	ACM					
	200 OK INVITE(ANM)	+		<b>←</b>	ANM					
	Conversation									
	INVITE(CPG, sendonly)	<b>→</b>		→	CPG(hold)					
	200 OK INVITE(recvonly)	+								
	ACK	<b>→</b>								
	INFO(LOP)	<b>→</b>		<b>→</b>	LOP(request)					
	200 OK INFO	+								
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active, CTNb)					
	200 OK INFO	+								
	INVITE(sendrecv)	<b>→</b>								
	200 OK INVITE(sendrecv)	+								
	ACK	<b>→</b>								
	BYE(REL)	<b>→</b>		<b>→</b>	REL					
	200 OK BYE(RLC)	+		+	RLC					

TP412015	SIP reference: RFC 3261 [4]	Q.73	ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)			
TSS reference	ISUP-SIP-ISUP/SS/ECT					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Facility message with generic notif					
	Verify that the SUT is able to transfer					
	transfer, alerting" and the service act					
	ISUP FAC in a SIP INFO request con					
SIP parameter	INFO: Content-Type: application/ISUI					
values	INFO: Content-Type: application/ISUI				the MIME body	
ISUP parameter	CPG: Event indicator=progress, Gene					
values	FAC: Generic notification=call transfe	r activ		er nu		
Comments	ISUP		SUT		SIP-I	
	IAM	→		<b>→</b>	INVITE(IAM)	
	ACM	<b>←</b>		<b>←</b>	180 Ringing(ACM)	
	ANM	<b>←</b>		+	200 OK INVITE(ANM)	
		С	onversation	)		
	CPG(hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)	
				<b>←</b>	200 OK INVITE(recvonly)	
				<b>→</b>	ACK	
	FAC(call transfer active, CTNb)	<b>→</b>		<b>→</b>	INFO(FAC)	
				<b>←</b>	200 OK INFO	
				<b>→</b>	INVITE(sendrecv)	
				+	200 OK INVITE(sendrecv)	
				<b>→</b>	ACK	
	REL	<b>→</b>		<b>→</b>	BYE(REL)	
	RLC	+		+	200 OK BYE(RLC)	

TP412016	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15				
TSS reference	ISUP-SIP-ISUP/SS/ECT							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Facility message with generic notification sent to the remote user  Verify that the SUT is able to transfer the generic notification generic notification set to  "call transfer, active" or "call transfer, alerting" and the service activation parameter set to  "call transfer" received in a SIP-I INFO request containing the ISUP FAC message in an  ISUP FAC message. O-MGCF interworking.							
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC encapsulated in the MIME body							
ISUP parameter	CPG: Event indicator=progre							
values	FAC: Generic notification=ca	all tran	sfer active, Cal	ll trar	nsfer number(PIXIT)			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		1	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation					
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	<b>→</b>						
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active, CTNb)			
	200 OK INFO	+						
	INVITE(sendrecv)	<b>→</b>						
	200 OK INVITE(sendrecv)	<b>←</b>						
	ACK	<b>→</b>						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP412017	SIP reference: RFC 3261 [4	]	Q.7	-	SUP reference: Q.1912.5 [1], i.5], clause 7.5.2.1.1.1 a)	
TSS reference	ISUP-SIP-ISUP/SS/ECT					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Call progress message with generic notification sent to the remote user  Verify that the transfer the CPG with the generic notification set to "call transfer, active" and the service activation parameter set to "call transfer" in a SIP-I INFO request containing the ISUP CPG message.  O-MGCF interworking.					
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body					
ISUP parameter values	CPG: Generic notification=call transf	er activ	e, Call tran	sfer n	umber (PIXIT)	
Comments	ISUP		SUT		SIP-I	
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	CPG(call transfer active, CTNb)	<b>→</b>		<b>→</b>	INFO(CPG)	
				+	200 OK INFO	
	ANM	+		+	200 OK INVITE(ANM)	
		С	onversation	n		
	REL	<b>→</b>		<b>→</b>	BYE(REL)	
	RLC	+		+	200 OK BYE(RLC)	

TP412018	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15					
TSS reference	ISUP-SIP-ISUP/SS/ECT	ISUP-SIP-ISUP/SS/ECT							
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Call progress message with generic notification sent to the remote user  Verify that the SUT is able to transfer the ISUP CPG with the generic notification set to  "call transfer, active" and the service activation parameter set to "call transfer" contained in SIP-I INFO request in an ISUP CPG. The held user is retrieved by receiving a re-INVITE sendrecv.  I-MGCF interworking.								
SIP parameter	INFO: Content-Type: applica	ation/IS	SUP; CPG en	capsu	lated in the MIME body				
values									
ISUP parameter	CPG: Event indicator=progre								
values	CPG: Generic notification=c	all trar		all tra					
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversatio	n					
	INVITE(CPG, sendonly)	→		→	CPG(hold)				
	200 OK INVITE(recvonly)	+							
	ACK	→							
	INFO(CPG)	→		→	CPG(call transfer active, CTNb)				
	200 OK INFO	+							
	INVITE(sendrecv)	→							
	200 OK INVITE(sendrecv)	+							
	ACK	<b>→</b>							
	BYE(REL)	→		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP412019	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15				
TSS reference	ISUP-SIP-ISUP/SS/ECT							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Call transfer number - removal of number							
					umber in the SIP-I INFO request			
					t exchange, if its indicator is set to			
					ement to transfer the number.			
SIP parameter	INFO: Content-Type: applica							
values		ation/IS	SUP; FAC(CTN	Nb=re	estricted) encapsulated in the MIME			
	body	body						
ISUP parameter	CPG: Event indicator=progre							
values	FAC: Generic notification=ca	all tran	sfer active, no	Call	transfer number(PIXIT)			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation					
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	→						
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active)			
	200 OK INFO	+						
	INVITE(sendrecv)	<b>→</b>						
	200 OK INVITE(sendrecv)	+						
	ACK	<b>→</b>						
		+-						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		<del>-</del>	RLC			

TP412020	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15					
TSS reference	ISUP-SIP-ISUP/SS/ECT								
SIP selection criteria									
ISUP selection									
criteria									
Test purpose	Call transfer number - con	versio	n to internation	onal	number				
		Verify that the IUT converts the <b>call transfer number</b> contained in the SIP-I INFO request into international format. The nature of address indicator shall be set to "international number".							
SIP parameter values	INFO: Content-Type: application	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=national) encapsulated in the MIME body							
ISUP parameter	CPG: Event indicator=progre								
values		all tran		ll trar	nsfer number=international(PIXIT)				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		<b>→</b>	IAM				
	180 Ringing(ACM)	+		<b>←</b>	ACM				
	200 OK INVITE(ANM)	+		<b>←</b>	ANM				
		Conversation							
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)				
	200 OK INVITE(recvonly)	+							
	ACK	<b>→</b>							
	INFO(FAC)	<b>→</b>		<b>→</b>	FAC(call transfer active, CTNb)				
	200 OK INFO	+							
	INVITE(sendrecv)	<b>→</b>							
	200 OK INVITE(sendrecv)	+							
	ACK	<b>→</b>							
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		<b>←</b>	RLC				

TP412021	SIP reference: RFC 3261 [4	]	Q.732	ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)			
TSS reference	ISUP-SIP-ISUP/SS/ECT						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Call transfer number - removal of own country code  Verify that the IUT removes the country code in the address signals of the call transfer number if it is the network's own country code contained in the ISUP FAC message. The nature of address indicator shall be set to "national (significant) number"						
SIP parameter							
values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAC(CTNb=international) encapsulated in the MIME body						
ISUP parameter	CPG: Event indicator=progress, Ger	eric not	ification=hold	1			
values	FAC: Generic notification=call transf	er active	e, Call transfe	r nu	ımber=national(PIXIT)		
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		<b>←</b>	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
		С	onversation				
	CPG(hold)	→		<b>→</b>	INVITE(CPG, sendonly)		
				<del>(</del>	200 OK INVITE(recvonly)		
				<b>→</b>	ACK		
	FAC(call transfer active, CTNb)	<b>→</b>		<b>→</b>	INFO(FAC)		
				<del>(</del>	200 OK INFO		
				<b>→</b>	INVITE(sendrecv)		
				<del>(</del>	200 OK INVITE(sendrecv)		
				<b>→</b>	ACK		
	REL	→		<u>→</u>	BYE(REL)		
	RLC	←		<del>(</del>	200 OK BYE(RLC)		

TP412022	SIP reference: RFC 3261 [4	Q.73	ISUP reference: Q.1912.5 [1], Q.732.7 [i.5], clause 7.5.2.1.1.1 a)				
TSS reference	ISUP-SIP-ISUP/SS/ECT						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Verify that if the IUT is able to transfer the sub-address in the access transport parameter in the ISUP FAC message contained in the SIP-I INFO request in ISUP FAC message and vice versa received in an ISUP FAC message in a SIP-I INFO request containing the ISUP FAC message. These are the calling sub-address for incoming calls and the connected sub-address for outgoing calls.  O-MGCF interworking.						
SIP parameter	INFO: Content-Type: application/ISL	JP: FAC	encapsulat	ed in	the MIME body		
values	in a comon type approauctives	, , , , , , ,	ooapoula		200,		
ISUP parameter	FAC: Generic notification=call transf	er active	e, Call trans	fer nu	ımber(PIXIT)		
values	FAC: ATP contained the connected				,		
Comments	ISUP		SUT		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+		+	200 OK INVITE(ANM)		
		C	onversatio	n			
	FAC(call transfer active, CTNb)	<b>→</b>		<b>→</b>	INFO(FAC)		
				+	200 OK INFO		
	FAC(ATP=SUB)	+		+	INFO(FAC)		
	,			<b>→</b>	200 OK INFO		
	FAC(ATP=SUB)	<b>→</b>		<b>→</b>	INFO(FAC)		
	·			+	200 OK INFO		
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		+	200 OK BYE(RLC)		

## 5.3.13 3PTY

TP413001	SIP reference: RFC 3261 [4]	Q	_	SUP reference: Q.1912.5 [1], ! [i.8], clause 2.4; 2.2.1				
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Served user initiates 3PTY  Verify that the served user with two active calls is located, can successfully join this call (remote held user) to a three-way conversation, and notify the implied remote party accordingly.  The IUT should transfer an ISUP CPG message with the generic notification indicator set to "conference established" in a SIP-I INFO request containing the ISUP CPG message. The event indicator in the CPG should be set to "progress":  1. setup a call to user B;  2. put this call on hold;  3. join this call to a conference.  O-MGCF interworking.							
SIP parameter		INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body						
values	or coment type: application, co. ; c.	o onoapouno		200,				
ISUP parameter	CPG: Event indicator=progress, Generic n	otification=h	old					
values	CPG: Event indicator=progress, Generic n			nce established				
Comments	ISUP	SUT		SIP-I				
	IAM →		<b>→</b>	INVITE(IAM)				
	ACM ←		<b>←</b>	180 Ringing(ACM)				
	ANM ←		←	200 OK INVITE(ANM)				
		Conversation						
	CPG(hold) →		→	INVITE(CPG, sendonly)				
			+	200 OK INVITE(recvonly)				
			<b>→</b>	ACK				
	CPG(conference established) →		<b>→</b>	INVITE(CPG, sendrecv)				
			<u>+</u>	200 OK INVITE(sendrecv)				
			<b>→</b>	ACK				
		Conversation						
	REL →		<b>→</b>	BYE(REL)				
	RLC +		<b>←</b>	200 OK BYE(RLC)				

TP413002	SIP reference: RFC	3261 [	4]	(	ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clause 2.4; 2.2.1			
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection criteria								
ISUP selection								
criteria								
Test purpose	Served user initiates 3PTY  Verify that the served user with two active calls is located, can successfully join this call (remote held user) to a three-way conversation, and notify the implied remote party accordingly.  The IUT should send a CPG message with the generic notification indicator set to "conference established" to both implied parties. The event indicator in the CPG should be set to "progress":  1. setup a call to user B;  2. put this call on hold;  3. join this call to a conference.  I-MGCF interworking							
SIP parameter	INFO: Content-Type: applica	tion/IS	UP; CPG e	ncapsu	lated in the MIME body			
values				·	·			
ISUP parameter	CPG: Event indicator=progre							
values	CPG: Event indicator=progre	ess, Ge		cation=				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		→	IAM			
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ACM			
	200 OK INVITE(ANM)	<b>←</b>		<b>←</b>	ANM			
			Conversation	on				
	INVITE(CPG, sendonly)	<b>→</b>		→	CPG(hold)			
	200 OK INVITE(recvonly)	+						
	ACK	<b>→</b>						
	INVITE(CPG, sendrecv)	<b>→</b>		<b>→</b>	CPG(conference established)			
	200 OK INVITE(sendrecv)	+						
	ACK →							
	Conversation							
	BYE(REL)	<b>→</b>		→	REL			
	200 OK BYE(RLC)	+		<b>←</b>	RLC			

TP413003	SIP reference: RFC 3261 [4]		Į,	SUP reference:				
			Q.1912.5 [1],					
		Q.7	34.2	[i.8], clause 2.4; 2.2.1				
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Served user initiates 3PTY  Verify that the served user with two active calls is located, can successfully join this call (remote active user) to a three-way conversation, and notify the implied remote party accordingly.  The IUT should send a CPG message with the generic notification indicator set to "conference established" to both implied parties. The event indicator in the CPG should be set to "progress":  1. setup a call to user B;  2. establish a conference.  O-MGCF interworking.							
SIP parameter values	INFO: Content-Type: application/ISUP; CF	G encapsulate	d in	the MIME body				
ISUP parameter values	CPG: Event indicator=progress, Generic n	otification=conf	erer	nce established				
Comments	ISUP	SUT		SIP-I				
	IAM →		<b>→</b>	INVITE(IAM)				
	ACM ←		<del>-</del>	180 Ringing(ACM)				
	ANM ←		<del>-</del>	200 OK INVITE(ANM)				
		Conversation						
	CPG(conference established) → INFO(CPG)							
			<b>←</b>	200 OK INFO				
		Conversation						
	REL →		<b>→</b>	BYE(REL)				
	RLC ←		<b>←</b>	200 OK BYE(RLC)				

TP413004	SIP reference: RF	C 3261 [	[4]	(	ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clause 2.4; 2.2.1			
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Served user initiates 3PTY  Verify that the served user with two active calls is located, can successfully join this call (remote active user) to a three-way conversation, and notify the implied remote party accordingly.  The IUT should send a CPG message with the generic notification indicator set to "conference established" to both implied parties. The event indicator in the CPG should be set to "progress":  1. setup a call to user B;  2. establish a conference.  I-MGCF interworking.							
SIP parameter values	INFO: Content-Type: appli	cation/IS	UP; CPG end	capsu	lated in the MIME body			
ISUP parameter values	CPG: Event indicator=proc	gress, Ge	eneric notifica	tion=	conference established			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
		<b>→</b>	Conversation	1				
	INFO(CPG)	<b>→</b>	CPG(conference established)					
	200 OK INFO	<b>←</b>						
			Conversation	1				
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP413005	SIP reference: RFC 3261 [4]	ISUP reference:									
			0.7		Q.1912.5 [1],						
TCC reference	IOLID OID IOLID/OO/ODTV		Q.73	34.2	[i.8], clause 2.5.2.1.1.3 a						
TSS reference SIP selection	ISUP-SIP-ISUP/SS/3PTY										
criteria											
ISUP selection											
criteria											
Test purpose	Served user creates a private communication with a remote user  Verify that a 3PTY call can successfully create private communication with the active-held user. The appropriate notification received in a ISUP CPG and is sent in INVITE/INFO (CPG) messages to the SIP-I.  O-MGCF interworking.										
SIP parameter values	INFO: Content-Type: application/ISUF	P; CPG	encapsulat	ed in	the MIME body						
ISUP parameter	CPG 1, 4: Event indicator=progress, 0										
values	CPG 5: Event indicator=progress, Ge										
		CPG 2: Event indicator=progress, Generic notification=conference established									
_	CPG 3: Event indicator=progress, Ge	neric n		onfer							
Comments	ISUP		SUT	<u> </u>	SIP-I						
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)						
	ACM	+		<del>-</del>	180 Ringing(ACM)						
	ANM ← 200 OK INVITE(ANM)										
	CPG 1(hold) → INVITE(CPG, sendon)										
	CPG 1(hold)	7		<b>→</b>	INVITE(CPG, sendonly)						
				<b>←</b>	200 OK INVITE(recvonly) ACK						
				7	ACK						
	CPG 2(conference established)	<b>→</b>		<b>→</b>	INVITE(CPG, sendrecv)						
	Ci d'alconierence established)	-		É	200 OK INVITE(sendrecv)						
				<b>→</b>	ACK						
				<del>-</del>	AOR						
	CPG 3(conference disconnected)	<b>→</b>		<b>→</b>	INFO(CPG)						
		+ -		<del>′</del>	200 OK INFO						
				<u> </u>	200 01(1111 0						
	CPG 4(hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)						
				+	200 OK INVITE(recvonly)						
				<b>→</b>	ACK						
	CPG 5(retrieve)	<b>→</b>		<b>→</b>	INVITE(CPG, sendrecv)						
	, ,			+	200 OK INVITE(sendrecv)						
				<b>→</b>	ACK						
		С	onversatio	n							
	CPG 6(conference established)	<b>→</b>		<b>→</b>	INFO(CPG)						
				+	200 OK INFO						
		_	onversatio								
	REL	<b>→</b>		<b>→</b>	BYE(REL)						
	RLC	+		<b>←</b>	200 OK BYE(RLC)						

TP413006	SIP reference: RFC	3261		ISUP reference:					
					Q.1912.5 [1],				
				Q	.734.2 [i.8], clause 2.5.2.1.1.3 a				
TSS reference	ISUP-SIP-ISUP/SS/3PTY								
SIP selection									
criteria									
ISUP selection									
criteria				141					
Test purpose	Served user creates a priva	ate co	ommunicati	on with	a remote user				
	Varify that a 2PTV call can s	110000	efully croate	nrivoto	communication with the active-held				
					E/INFO (CPG) and is sent in <b>CPG</b>				
	messages to the ISUP.	ation	10001100 111 0	<b>.</b>					
	I-MGCF interworking.								
SIP parameter	INFO: Content-Type: applica	tion/I	SUP: CPG e	encapsu	lated in the MIME body				
values	7, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1, 1,		<b>,</b>		,				
ISUP parameter	CPG: Event indicator=progre	ess, G	eneric notifi	cation=	hold				
values	CPG: Event indicator=progre								
	CPG: Event indicator=progress, Generic notification=conference established								
	CPG: Event indicator=progre	ess, G		cation=					
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	→		<b>→</b>	IAM				
	180 Ringing(ACM)	+		<b>←</b>	ACM				
	200 OK INVITE(ANM)	<b>←</b>		<b>←</b>	ANM				
	Conversation								
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)				
	200 OK INVITE(recvonly)	+							
	ACK	<b>→</b>							
	N. ((TE (ODO )	+_			000/ (				
	INVITE(CPG, sendrecv)	<b>→</b>		→	CPG(conference established)				
	200 OK INVITE(sendrecv)	<b>+</b>							
	ACK	<b>→</b>	1						
	INEO(CDC)	<b>→</b>			CDC(somforence discourse stad)				
	INFO(CPG) 200 OK INFO	<del>7</del>		<b>→</b>	CPG(conference disconnected)				
	ZUU UK IINFU	+							
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)				
	200 OK INVITE(recvonly)	+		<del>                                     </del>	Or O(riola)				
	ACK	<b>→</b>							
	7.01	+							
	INVITE(CPG, sendrecv)	<b>→</b>		<b>→</b>	CPG(retrieve)				
	200 OK INVITE(sendrecv)	+			J. 2(1011010)				
	ACK	<b>→</b>							
			Conversati	on					
	INFO(CPG)	<b>→</b>		<u>∵</u>	CPG(conference established)				
	200 OK INFO	+							
		† -	Conversati	on					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				
	1200 OK DIE(KLC)			7	INLU				

TP413007	SIP reference: RFC 3261 [4	ij	Q.73	ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clause 2.5.2.1.1.3 a				
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Served user creates a private communication with a remote user  Verify that the IUT (controlling the conference) on a 3PTY call can successfully create private communication with the active-idle user. The appropriate notification is sent in CPG messages to the user.  O-MGCF interworking.							
SIP parameter values	INFO: Content-Type: application/ISUP; CPG encapsulated in the MIME body							
ISUP parameter	CPG: Event indicator=progress, Ger	neric noti	fication=con	ıferei	nce established			
values	CPG: Event indicator=progress, Ger	neric noti	fication=con	ıferei	nce disconnected			
Comments	ISUP		SUT		SIP-I			
	IAM	→		<b>→</b>	INVITE(IAM)			
	ACM	<b>←</b>		<b>←</b>	180 Ringing(ACM)			
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)			
		С	onversatior	1				
	CPG(conference established)	→		<b>→</b>	INFO(CPG)			
				<b>←</b>	200 OK INFO			
	CPG(conference disconnected)	→		<b>→</b>	INFO(CPG)			
				<b>←</b>	200 OK INFO			
			onversatior	1				
	CPG(conference established)	→		<b>→</b>	INFO(CPG)			
				<b>←</b>	200 OK INFO			
			onversatior	1				
	REL	→		<b>→</b>	BYE(REL)			
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)			

TP413008	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.734.2 [i.8], clause 2.5.2.1.1.3 a				
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Served user creates a private communication with a remote user  Verify that the IUT (controlling the conference) on a 3PTY call can successfully create private communication with the active-idle user. The appropriate notification is sent in CPG messages to the user.  I-MGCF interworking.							
SIP parameter	INFO: Content-Type: appli	cation/IS	SUP; CPG enc	apsı	lated in the MIME body			
values				-	·			
ISUP parameter	CPG: Event indicator=prog	gress, G	eneric notificat	ion=	conference established			
values	CPG: Event indicator=prog	gress, G	eneric notificati	ion=				
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	→		<b>^</b>	IAM			
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ACM			
	200 OK INVITE(ANM)	<b>←</b>		<b>←</b>	ANM			
			Conversation					
	INFO(CPG)	→		<b>→</b>	CPG(conference established)			
	200 OK INFO	<b>←</b>						
	INFO(CPG)	→		<b>→</b>	CPG(conference disconnected)			
	200 OK INFO	<b>←</b>						
			Conversation					
	INFO(CPG)	<b>→</b>		<b>^</b>	CPG(conference established)			
	200 OK INFO	<b>←</b>						
			Conversation					
	BYE(REL)	<b>→</b>		1	REL			
	200 OK BYE(RLC)	+		<b>4</b>	RLC			

TP413009	SIP reference: RFC 3261 [4]		Q.73	-	SUP reference: Q.1912.5 [1], i.8], clause 2.5.2.1.1.3 b			
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Served user disconnects one remote user and retains the other  Verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect the active-held user and retain and notify the other user appropriately using CPG messages.  The IUT should send to the appropriate remote users CPG messages with a generic notification indicator. The event indicator in the CPG should be set to "progress".  O-MGCF interworking.							
SIP parameter	INFO: Content-Type: application/ISUP;	CPG	encansulat	ed in	the MIME hody			
values	in o. contone Type, application/rect	0. 0	orioapodiat	ou	the Minie Body			
ISUP parameter	CPG: Event indicator=progress, Gener	ic not	ification=cor	nferer	nce established			
values	CPG: Event indicator=progress, Gener							
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		<b>←</b>	180 Ringing(ACM)			
	ANM	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)			
		С	onversatio	n				
	CPG(conference established)	<b>→</b>		<b>→</b>	INFO(CPG)			
				<b>←</b>	200 OK INFO			
	CPG(conference disconnected)	<b>→</b>		<b>→</b>	INFO(CPG)			
				<b>←</b>	200 OK INFO			
		С	onversatio	n				
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	<b>←</b>		<b>←</b>	200 OK BYE(RLC)			

TP413010	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], .734.2 [i.8], clause 2.5.2.1.1.3 b			
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Served user disconnects one remote user and retains the other  Verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect the active-held user and retain and notify the other user appropriately using CPG messages.  The IUT should send to the appropriate remote users CPG messages with a generic notification indicator. The event indicator in the CPG should be set to "progress" I-MGCF interworking.							
SIP parameter values	INFO: Content-Type: applica	tion/IS	SUP; CPG enca	apsu	lated in the MIME body			
ISUP parameter	CPG: Event indicator=progre	ss, G	eneric notificati	on=	conference established			
values	CPG: Event indicator=progre	ss, G	eneric notificati	on=	conference disconnected			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		<b>←</b>	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
			Conversation					
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference established)			
	200 OK INFO	+						
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference disconnected)			
	200 OK INFO	+						
			Conversation					
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP413011	SIP reference: RFC 3261 [4]			ı	SUP reference:			
					Q.1912.5 [1],			
			Q.73	34.2 [	i.8], clause 2.5.2.1.1.3 b			
TSS reference	ISUP-SIP-ISUP/SS/3PTY							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Carved user disconnects one remot	0 1100	r and ratain	o the	othor			
rest purpose	Served user disconnects one remote user and retains the other  Verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect the active-idle user and retain and notify the other user appropriately using CPG messages.  The IUT should send to the appropriate remote users CPG messages with a generic notification indicator. The event indicator in the CPG should be set to "progress".							
SIP parameter	O-MGCF interworking.  INFO: Content-Type: application/ISUP	· CPG	encansulate	ed in	the MIME body			
values	3. Contont Type. application/1001	, 01 0	onoupsulati	ou III	and whive body			
ISUP parameter	CPG: Event indicator=progress, Gene	ric not	ification=hole	d				
values	CPG: Event indicator=progress, Gene				nce established			
	CPG: Event indicator=progress, Gene	ric not	ification=con	nferer				
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		<b>←</b>	200 OK INVITE(ANM)			
			onversation	1				
	CPG(hold)	<b>→</b>		<b>→</b>	INVITE(CPG, sendonly)			
				+	200 OK INVITE(recvonly)			
				<b>→</b>	ACK			
	CPG(conference established)	<b>→</b>		<b>→</b>	INVITE(CPG, sendrecv)			
				+	200 OK INVITE(sendrecv)			
				<b>→</b>	ACK			
		<u> </u>						
	CPG(conference disconnected)	<b>→</b>		<b>→</b>	INFO(CPG)			
				+	200 OK INFO			
	CDC(hold)	<b>→</b>		_	INIVITE/CDC condonic			
	CPG(hold)	7		<b>→</b>	INVITE(CPG, sendonly) 200 OK INVITE(recvonly)			
				<b>→</b>	ACK			
				7	AUN			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	<del>-</del>		<del>7</del>	200 OK BYE(RLC)			
	INLU	_		_	ZUU ON DIE(NLU)			

TP413012	SIP reference: RFC	3261		ISUP reference:						
					Q.1912.5 [1],					
		Q.734								
TSS reference	ISUP-SIP-ISUP/SS/3PTY									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	Verify that the IUT (controllin the active-idle user and retain messages. The IUT should send to the a	The IUT should send to the appropriate remote users CPG messages with a generic notification indicator. The event indicator in the CPG should be set to "progress".								
SIP parameter	INFO: Content-Type: applica	tion/I		ncancu	lated in the MIME body					
values	in C. Content-Type, applica	uoi/i	, OI G 6	σποαρδυ	iated in the Milvic body					
ISUP parameter	CPG: Event indicator=progre	ess, G	eneric notifi	cation=l	nold					
values	CPG: Event indicator=progre									
	CPG: Event indicator=progre	ess, G		cation=						
Comments	SIP-I		SUT		ISUP					
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM					
	180 Ringing(ACM)	+		<del>-</del>	ACM					
	200 OK INVITE(ANM)	+	<u> </u>	<b>←</b>	ANM					
	W W ### (0.50	Conversation								
	INVITE(CPG, sendonly)	<b>→</b>		→	CPG(hold)					
	200 OK INVITE(recvonly)	<b>←</b>								
	ACK	7								
	INVITE(CPG, sendrecv)	<b>→</b>		→	CPG(conference established)					
	200 OK INVITE(sendrecv)	<del>′</del>			Or O(contenence established)					
	ACK	<b>→</b>								
	INFO(CPG)	<b>→</b>		<b>→</b>	CPG(conference disconnected)					
	200 OK INFO	+								
	INVITE(CPG, sendonly)	<b>→</b>		<b>→</b>	CPG(hold)					
	200 OK INVITE(recvonly)	+								
	ACK	→								
	DVE(DEL)	+	1		DEL					
	BYE(REL)	<b>→</b>		<b>→</b>	REL					
	200 OK BYE(RLC)	7	L		RLC					

## 5.3.14 User-to-user service

### 5.3.14.1 User-to-user service 1

TP414001	SIP reference: RFC 3261 [4] ISUP reference:								
				Q.1912	2.5 [1], clauses A.1.1,				
					2.3 and 4, Q.737 [i.10]				
TSS reference	ISUP-SIP-ISUP/SS/UUS1								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Service 1 implicit request: U	Jser-to-	user information in	the IAN	1				
				er-to-us	er service 1 implicit request in				
	the encapsulated IAM. O-M								
SIP parameter	INVITE: Content-Type: app	lication	/ISUP; IAM contain	ing the	user-to-user information				
values	parameter encapsulated in	the MIN	ЛE body						
ISUP parameter	IAM: User-to-user information	on para	meter						
values									
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM ← 200 OK INVITE(ANM)								
			Conversation						
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP414002	SIP reference: RF	C 3261	[4]	Q.191	SUP reference: 2.5 [1], clauses A.1.1 2.3 and 4, Q.737 [i.10]				
TSS reference	ISUP-SIP-ISUP/SS/UUS1		L						
SIP selection criteria									
ISUP selection criteria									
Test purpose	Service 1 implicit request: User-to-user information in the INVITE  Ensure that the SUT can successfully transfer the User-to-user service 1 implicit request in the encapsulated IAM. I-MGCF interworking.								
SIP parameter values	INVITE: Content-Type: appragneter encapsulated in	olication	/ISUP; IAM cor	ntaining the	user-to-user information				
ISUP parameter values	IAM: User-to-user informat								
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)	+		+	ANM				
			Conversation	on					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

TP414003	SIP reference	e: RFC 3261 [4	]	Q.191	SUP reference: 2.5 [1], clauses A.1.1, 2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/U	US1			
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT of not essential in the en		,		er service 1 explicit request
SIP parameter values	INVITE: Content-Type parameter encapsulat			ining the	user-to-user indicator
ISUP parameter values	IAM: User-to-user info	ormation paran	neter, User-to-us	ser indica	tor = service 1 explicit
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversation		
	REL	<b>→</b>	•	<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP414004	SIP reference: RFC	SIP reference: RFC 3261 [4] ISUP reference:							
				Q.	.1912	2.5 [1], clauses A.1.1,			
				1.	1.5.2	.3 and 4, Q.737 [i.10]			
TSS reference	ISUP-SIP-ISUP/SS/UUS1								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Service 1 explicit request: U	Jser-to-	user indica	tor in the II	NVIT	E			
SIP parameter	Ensure that the SUT can successfully transfer the User-to-user service 1 explicit request essential received in the IAM. O-MGCF interworking.  INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator								
values	parameter encapsulated in 180 Ringing: Content-Type:			ACM cont	ainin	a the user-to-user indicator			
	parameter encapsulated in			/ CON CON		g the deer to deer maleater			
ISUP parameter	IAM: User-to-user information	on para	ameter, Use	r-to-user ir	ndica	tor			
values	ACM: User-to-user indicator	r set to	service 1 s	upported r	espo	nse			
Comments	ISUP		SUT			SIP-I			
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ACM	+			+	180 Ringing(ACM)			
	ANM	+			+	200 OK INVITE(ANM)			
			Convers	ation					
	REL	<b>→</b>			<b>→</b>	BYE(REL)			
	RLC	<b>←</b>			+	200 OK BYE(RLC)			

TP414005	SIP reference: RFC	3261	[4]		Į.	SUP reference:
				Q.	.1912	2.5 [1], clauses A.1.1,
				1.	1.5.2	2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Service 1 explicit request: U	lser-to-	user indicat	or in the II	VVIT	E
	Ensure that the SUT can su					
	essential received in the end					
SIP parameter	INVITE: Content-Type: appl			containing	the	user-to-user indicator
values	parameter encapsulated in t					
	180 Ringing: Content-Type:	applica	ation/ISUP;	ACM cont	ainin	g the user-to-user indicator
	parameter encapsulated in t					
ISUP parameter	IAM: User-to-user information					
values	ACM: User-to-user indicator	set to			espo	
Comments	SIP-I		SUT			ISUP
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM
	180 Ringing(ACM)	+			<b>←</b>	ACM
	200 OK INVITE(ANM)	<b>+</b>			+	ANM
			Convers	ation		
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	<b>+</b>			<b>←</b>	RLC

TP414006	SIP reference: RFC	3261	[4]		I;	SUP reference:
		'			2.1912	2.5 [1], clauses A.1.1,
						.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1					
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose		ccessf	ully transfer	the User	-to-us	er service 1 implicit response
	in the encapsulated ACM.					
	O-MGCF interworking.					
SIP parameter	Service 1 implicit response:	User-te	o-user infor	mation in	the 18	30 Ringing
values						
	INVITE: Content-Type: appl			containin	g the	user-to-user information
	parameter encapsulated in t					
				ACM cor	ntainin	g the user-to-user information
	parameter encapsulated in t	the MIN	ЛE body			
ISUP parameter	IAM: User-to-user information					
values	ACM: User-to-user informat	ion par	ameter			
Comments	ISUP		SUT			SIP-I
	IAM	<b>↑</b>			<b>→</b>	INVITE(IAM)
	ACM	+			+	180 Ringing(ACM)
	ANM	+			+	200 OK INVITE(ANM)
			Convers	ation		
	REL	<b>→</b>			<b>→</b>	BYE(REL)
	RLC	+			+	200 OK BYE(RLC)

TP414007	SIP reference: RF	C 3261	[4]		ISUP reference:
					2.5 [1], clauses A.1.1, 2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1				.,
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 1 implicit response	: User-t	o-user infor	mation in the A	ACM
SIP parameter values	in the encapsulated ACM. I-MGCF interworking. INVITE: Content-Type: appraameter encapsulated in 180 Ringing: Content-Type	olication, the MIN	/ISUP; IAM /IE body ation/ISUP;	containing the	ser service 1 implicit response user-to-user information ng the user-to-user information
ICLID navamatar	parameter encapsulated in				
ISUP parameter values	IAM: User-to-user informat ACM: User-to-user informat				
Comments	SIP-I	lition par	SUT	r	ISUP
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM
	180 Ringing(ACM)	+		<del>(</del>	ACM
	200 OK INVITE(ANM)	+		+	ANM
			Convers		
	BYE(REL)	<b>→</b>		→	REL
	200 OK BYE(RLC)	<b>←</b>		←	RLC

TP414008	SIP reference: RFC	3261	[4]	Q.191	ISUP reference: 2.5 [1], clauses A.1.1, 2.3 and 4, Q.737 [i.10]			
TSS reference	ISUP-SIP-ISUP/SS/UUS1							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Service 1 explicit response service 1 not supported in the 180 Ringing  Ensure that the SUT can successfully transfer the User-to-user service 1 explicit response not supported in the encapsulated ACM.  O-MGCF interworking.							
SIP parameter values	INVITE: Content-Type: appl parameter encapsulated in t 180 Ringing: Content-Type: parameter encapsulated in t	the MIN applica	/IE body ation/ISUP; A	· ·	user-to-user information			
ISUP parameter	IAM: User-to-user information			o-user indica	ator set to service 1 request			
values	ACM: User-to-user indicator	set to	service 1 not	supported re	esponse			
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversat	ion				
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		+	200 OK BYE(RLC)			

TP414009	SIP reference: RFC	3261	[4]		-	SUP reference:
						2.5 [1], clauses A.1.1, 2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1					
SIP selection						
criteria						
ISUP selection criteria						
Test purpose	Service 1 explicit response	service	1 not supp	orted in th	ne AC	M
	not supported in the ACM. I-MGCF interworking.					er service 1 explicit response
SIP parameter	INVITE: Content-Type: appl			containin	g the	user-to-user information
values	parameter encapsulated in					
	180 Ringing: Content-Type: parameter encapsulated in the			ACM con	tainin	g the user-to-user indicator
ISUP parameter	IAM: User-to-user information					
values	ACM: User-to-user indicator	set to	service 1 ne	ot suppor	ted re	
Comments	SIP-I		SUT			ISUP
	INVITE(IAM)	<b>→</b>			<b>→</b>	IAM
	180 Ringing(ACM)	+			<del>-</del>	ACM
	200 OK INVITE(ANM)	<del>-</del>			+	ANM
			Convers	ation		
	BYE(REL)	<b>→</b>			<b>→</b>	REL
	200 OK BYE(RLC)	+			+	RLC

TP414010	SIP reference: RFC	3261	[4]	Q.191	ISUP reference: 2.5 [1], clauses A.1.1, 2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Ensure that the SUT can su network in the encapsulated O-MGCF interworking.			User-to-us	ser service 1 discarded by the
SIP parameter	INVITE: Content-Type: appl			aining the	user-to-user information
values	parameter encapsulated in a 180 Ringing: Content-Type: parameter encapsulated in a	applica	ation/ISUP; ACI	Л containir	ng the User-to-user indicator
ISUP parameter	IAM: User-to-user information				
values	ACM: User-to-user indicator	r set to	discarded by th	e network	response
Comments	ISUP		SUT		SIP-I
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)
	ACM	+		+	180 Ringing(ACM)
	ANM	+		+	200 OK INVITE(ANM)
			Conversatio	n	
	REL	<b>→</b>		<b>→</b>	BYE(REL)
	RLC	+		+	200 OK BYE(RLC)

TP414011	SIP reference: RFC	ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.1.5.2.3 and 4, Q.737 [i.10]							
TSS reference	ISUP-SIP-ISUP/SS/UUS1		J.		, , ,				
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT can sunetwork in the encapsulated I-MGCF interworking.			e User-to-us	er service 1 discarded by the				
SIP parameter	INVITE: Content-Type: app	lication	ISUP; IAM co	ntaining the	user-to-user information				
values	parameter encapsulated in								
	180 Ringing: Content-Type: parameter encapsulated in			CM containin	g the User-to-user indicator				
ISUP parameter	IAM: User-to-user information	on para	meter						
values	ACM: User-to-user indicato	r set to	discarded by t	he network	response				
Comments	SIP-I		SUT		ISUP				
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM				
	180 Ringing(ACM)	+		+	ACM				
	200 OK INVITE(ANM)								
			Conversati	on					
	BYE(REL)	<b>→</b>		<b>→</b>	REL				
	200 OK BYE(RLC)	+		+	RLC				

# 5.3.14.2 User-to-user service 2

TP414101	SIP reference: RF0	C 3261	[4]	ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.2.5.2.3 and 4, Q.737 [i.10]					
TSS reference	ISUP-SIP-ISUP/SS/UUS2								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	Ensure that the SUT can su and User-to user information	Service 2 request not essential transferred in the INVITE  Ensure that the SUT can successfully transfer the User-to-user service 2 explicit request and User-to user information in the encapsulated IAM. An additional User-to-user information is sent in a USR message encapsulated in an INFO request.							
SIP parameter	INVITE: Content-Type: app	lication	/ISUP: IAM o	containing the	user-to-user indicator and				
values	User-to-user information er INFO: Content-Type: applic parameter encapsulated in	ncapsula cation/IS	ated in the M SUP; USR co	IIME body					
ISUP parameter	IAM: User-to-user informati			-to-user indica	ator				
values	USR: User-to-user informati			to acci maior					
Comments	ISUP		SUT		SIP-I				
	IAM	<b>→</b>		→	INVITE(IAM)				
	ACM	+		+	180 Ringing(ACM)				
	ANM	+		+	200 OK INVITE(ANM)				
			Conversa	tion	<u> </u>				
	USR	<b>→</b>		<b>→</b>	INFO(USR)				
				+	200 OK INFO				
	USR	+		+	INFO(USR)				
				<b>→</b>	200 OK INFO				
	REL	<b>→</b>		<b>→</b>	BYE(REL)				
	RLC	+		+	200 OK BYE(RLC)				

TP414102	SIP reference: R	FC 3261 [4	]	Q.191	ISUP reference: 2.5 [1], clauses A.1.1, 2.3 and 4, Q.737 [i.10]			
TSS reference	ISUP-SIP-ISUP/SS/UUS2	2						
SIP selection criteria								
ISUP selection criteria								
Test purpose		successfull additional	y transfer the l	Jser-to-u	ser service 2 explicit request in n is sent in a USR message			
SIP parameter values	INVITE: Content-Type: ap encapsulated in the MIME INFO: Content-Type: app parameter encapsulated	E body lication/ISU in the MIME	JP; USR contains	ining the	User-to-user information			
ISUP parameter values	IAM: User-to-user information USR: User-to-user information user informati	ation param	eter, User-to-u	user indic	ator			
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	·		Conversation	)				
	INFO(USR)	<b>→</b>		→	USR			
	200 OK INFO ←							
	INFO(USR)	+		+	USR			
	200 OK INFO	<b>→</b>						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	+		+	RLC			

TP414103	SIP reference: RFC	3261	[4]	Q.191	SUP reference: 2.5 [1], clauses A.1.1, 2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 2 response not prov	/ided tr	ansferred in the I	NVITE	
SIP parameter values	not supported in the encaps I-MGCF interworking. INVITE: Content-Type: appl parameter encapsulated in	ication,	ACM. (ISUP; IAM conta (IE body	ining the	user-to-user information  g the user-to-user indicator
	parameter encapsulated in				
ISUP parameter	IAM: User-to-user information				
values	ACM: User-to-user indicator	set to		ported re	
Comments	ISUP	<b>→</b>	SUT	<b>→</b>	SIP-I
	IAM ACM	<del>7</del>		<del>7</del>	INVITE(IAM)
	ANM	<del></del>		<del>-</del>	180 Ringing(ACM) 200 OK INVITE(ANM)
	AINIVI	_	Conversation		200 OK INVITE(ANIVI)
	REL	<b>→</b>	Jonversation	→	BYE(REL)
	RLC	+		<del>/</del>	200 OK BYE(RLC)

TP414104	SIP reference: RF	C 3261	[4]	Q.191	SUP reference: 2.5 [1], clauses A.1.1,
				1.2.5.2	2.3 and 4, Q.737 [i.10]
TSS reference	ISUP-SIP-ISUP/SS/UUS2				
SIP selection criteria					
ISUP selection criteria					
Test purpose	Service 2 response not pro	vided tr	ansferred in t	ne IAM	
SIP parameter values ISUP parameter values	Ensure that the SUT can s not provided in the encaps INVITE: Content-Type: app parameter encapsulated in IAM: User-to-user informat	ulated A olication the MIN	ACM. I-MGCF /ISUP; IAM co ME body	interworking ontaining the	user-to-user indicator
Comments	SIP-I		SUT		ISUP
	INVITE(IAM)	<b>→</b>		→	IAM
	180 Ringing(ACM)	+		+	ACM
	200 OK INVITE(ANM)	+		+	ANM
	` ` `		Conversat	ion	
	BYE(REL)	<b>→</b>		<b>→</b>	REL
	200 OK BYE(RLC)	+		+	RLC

#### 5.3.14.3 User-to-user service 3

TP414201	SIP reference: RF0	3261	[4]	Q.191	ISUP reference: 2.5 [1], clauses A.1.1,		
TSS reference	ISUP-SIP-ISUP/SS/UUS3			1.3.5.	2.3 and 4, Q.737 [i.10]		
SIP selection	150P-SIP-150P/55/0053						
criteria							
ISUP selection							
criteria							
Test purpose		itional l			ser service 3 explicit request in sent in several USR message		
SIP parameter	INVITE: Content-Type: app		/ISUP; IAM	containing the	user-to-user indicator		
values	INFO: Content-Type: applic parameter encapsulated in	encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body					
ISUP parameter	IAM: User-to-user informati		ameter, Use	r-to-user indica	ator		
values	USR: User-to-user informat	ion	_		_		
Comments	ISUP		SU'		SIP-I		
	IAM	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ACM	+		+	180 Ringing(ACM)		
	ANM	+	L	<b>+</b>	200 OK INVITE(ANM)		
			Convers		11120(1122)		
	USR	<b>→</b>		<b>→</b>	INFO(USR)		
				+	200 OK INFO		
	USR	+		+	INFO(USR)		
	0011			<b>→</b>	200 OK INFO		
					200 01(11(1)		
	USR	+		+	INFO(USR)		
				<b>→</b>	200 OK INFO		
	USR	<b>→</b>		<b>→</b>	INFO(USR)		
				+	200 OK INFO		
	REL	<b>→</b>		<b>→</b>	BYE(REL)		
	RLC	+		<b>←</b>	200 OK BYE(RLC)		

TP414202	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.3.5.2.3 and 4, Q.737 [i.10]			
TSS reference	ISUP-SIP-ISUP/SS/UUS3	3					
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose		ditional l			ser service 3 explicit request in sent in several USR message		
SIP parameter values	INVITE: Content-Type: ap encapsulated in the MIME INFO: Content-Type: app parameter encapsulated i	INVITE: Content-Type: application/ISUP; IAM containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body					
ISUP parameter values	IAM: User-to-user information USR: User-to-user information user informati		ımeter, Use	r-to-user indica	ator		
Comments	SIP-I		SU	Γ	ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
			Convers	ation			
	INFO(USR)	<b>→</b>		<b>→</b>	USR		
	200 OK INFO	+					
	INFO(USR)	+		+	USR		
	200 OK INFO	<b>→</b>					
	INFO(USR)	+		+	USR		
	200 OK INFO	<b>→</b>					
	INFO(USR)	<b>→</b>		<b>→</b>	USR		
	200 OK INFO	+					
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

TP414203	SIP reference: RF	C 3261	[4]	Q.191	ISUP reference: 2.5 [1], clauses A.1.1, 2.3 and 4, Q.737 [i.10]			
TSS reference	ISUP-SIP-ISUP/SS/UUS3							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages.  O-MGCF interworking.							
SIP parameter	INFO: Content-Type: applic		SUP; FAR conta	aining the ι	user-to-user indicator			
values	INFO: Content-Type: applie encapsulated in the MIME INFO: Content-Type: applie	encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAA containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information parameter encapsulated in the MIME body						
ISUP parameter	FAR: User-to-user indicato	r service	e 3 request not	essential				
values	FAA: User-to-user indicato	r service	3 response p	rovided				
	USR: User-to-user informa	tion						
Comments	ISUP		SUT		SIP-I			
	IAM	<b>→</b>		→	INVITE(IAM)			
	ACM	+		+	180 Ringing(ACM)			
	ANM	+		+	200 OK INVITE(ANM)			
			Conversation	on				
	FAR	<b>→</b>		<b>→</b>	INFO(FAR)			
				+	200 OK INFO			
	FAA	+		<b>←</b>	INFO(FAA)			
				<b>→</b>	200 OK INFO			
	USR	<b>→</b>		<b>→</b>	INFO(USR)			
				+	200 OK INFO			
	USR	+		+	INFO(USR)			
				<b>→</b>	200 OK INFO			
	USR	+		+	INFO(USR)			
				<b>→</b>	200 OK INFO			
	USR	<b>→</b>		<b>→</b>	INFO(USR)			
		<del>                                     </del>		<del>′</del>	200 OK INFO			
		1			200 010 1141 0			
	REL	<b>→</b>		<b>→</b>	BYE(REL)			
	RLC	+		<del>-</del>	200 OK BYE(RLC)			
	INLU			7	1200 ON DIE(KLC)			

TP414204	SIP reference: RF	C 3261	[4]		ISUP reference: 2.5 [1], clauses A.1.1, 2.3 and 4, Q.737 [i.10]			
TSS reference	ISUP-SIP-ISUP/SS/UUS3			1.3.3.	2.5 and 4, Q.757 [1.10]			
SIP selection	130F-31F-130F/33/0033							
criteria								
ISUP selection								
criteria								
Test purpose	confirmed state can succe	Ensure that a User-to-user request service 3 encapsulated in an INFO request during the confirmed state can successful proceeded. The User-to-user information is passed on in several encapsulated USR messages.						
SIP parameter values	INFO: Content-Type: appli encapsulated in the MIME INFO: Content-Type: appli encapsulated in the MIME	INFO: Content-Type: application/ISUP; FAR containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; FAA containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; USR containing the User-to-user information						
ISUP parameter values	FAR: User-to-user indicator FAA: User-to-user indicator USR: User-to-user information	or service or service	e 3 request n					
Comments	SIP-I		SUT		ISUP			
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM			
	180 Ringing(ACM)	+		+	ACM			
	200 OK INVITE(ANM)	+		+	ANM			
	200 311 11 11 12 (7 11 11 11 )	1	Conversa		7 (1 (1))			
	INFO(FAR)	<b>→</b>	Jonversa	<b>→</b>	FAR			
	200 OK INFO	É			IAK			
	200 010 1141 0	_						
	INFO(FAA)	+		+	FAA			
	200 OK INFO	<b>→</b>			1700			
	200 OK INFO	+ -						
	INFO(USR)	<b>→</b>		<b>→</b>	USR			
	200 OK INFO	<del>-</del>			0011			
	200 010 1141 0	_						
	INFO(USR)	+		+	USR			
	200 OK INFO	<b>→</b>			0011			
	200 OK IIVI O	+ -						
	INFO(USR)	+		+	USR			
	200 OK INFO	<b>→</b>						
	200 OK IIVI O							
	INFO(USR)	<b>→</b>		<b>→</b>	USR			
	200 OK INFO	+		<del>-                                     </del>				
	200 010 1141 0	+ •						
	BYE(REL)	<b>→</b>		<b>→</b>	REL			
	200 OK BYE(RLC)	<del>-</del>		· ·	RLC			
	1200 ON DIE(KLU)			7	INLO			

TP414205	SIP reference: RFC	3261	[4]	ISUP reference: Q.1912.5 [1], clauses A.1.1, 1.3.5.2.3 and 4, Q.737 [i.10]				
TSS reference	ISUP-SIP-ISUP/SS/UUS3							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Ensure that the SUT can su in the encapsulated ANM. O-MGCF interworking.	ccessf	ully transfer t	he User-t	o-us	er service 3 explicit response		
SIP parameter	INVITE: Content-Type: appl	lication	/ISUP; IAM c	ontaining	the	user-to-user indicator		
values	parameter encapsulated in	the MIN	∕IE body					
	200 OK INVITE: Content-Ty indicator parameter encaps				onta	ining the user-to-user		
ISUP parameter	IAM: User-to-user indicator			•				
values	ANM: User-to-user indicator				spons	se		
Comments	ISUP		SUT			SIP-I		
	IAM	<b>→</b>			<b>→</b>	INVITE(IAM)		
	ACM	+			<del>(</del>	180 Ringing(ACM)		
	ANM	<del>-</del>			<del>(</del>	200 OK INVITE(ANM)		
			Conversa	tion		Ì		
	REL	<b>→</b>			<b>→</b>	BYE(REL)		
	RLC	+			<b>←</b>	200 OK BYE(RLC)		

TP414206	SIP reference: RFC	3261	[4]		.1912	SUP reference: 2.5 [1], clauses A.1.1, .3 and 4, Q.737 [i.10]		
TSS reference	ISUP-SIP-ISUP/SS/UUS3							
SIP selection criteria								
ISUP selection								
criteria								
Test purpose		ccessf	ully transfer t	he User-	to-use	er service 3 explicit response		
	in the encapsulated ANM.							
oin .	O-MGCF interworking.		// CLUB 1414					
SIP parameter	INVITE: Content-Type: appl			ontainin	g the i	user-to-user indicator		
values	parameter encapsulated in							
	200 OK INVITE: Content-Ty				contai	ining the user-to-user		
	indicator parameter encaps							
ISUP parameter	IAM: User-to-user indicator							
values	ANM: User-to-user indicator	r set to	service 3 pro	ovided re	spons	se		
Comments	SIP-I		SUT			ISUP		
	INVITE(IAM)	→			<b>→</b>	IAM		
	180 Ringing(ACM)	+			+	ACM		
	200 OK INVITE(ANM) ← ANM							
			Conversa	tion				
	BYE(REL)	<b>→</b>			<b>→</b>	REL		
	200 OK BYE(RLC)	+			+	RLC		

TP414207	SIP reference: RF0	C 3261	[4]	Q.191	SUP reference: 2.5 [1], clauses A.1.1, 2.3 and 4, Q.737 [i.10]	
TSS reference	ISUP-SIP-ISUP/SS/UUS3					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Ensure that a User-to-user confirmed state can success				an INFO request during the request is rejected.	
SIP parameter values	INFO: Content-Type: application/ISUP; FAR containing the user-to-user indicator encapsulated in the MIME body INFO: Content-Type: application/ISUP; FRJ containing the user-to-user indicator encapsulated in the MIME body					
ISUP parameter	FAR: User-to-user indicato		e 3 request	not essential		
values	FRJ: User-to-user indicator					
Comments	ISUP		SU	Γ	SIP-I	
	IAM	<b>→</b>		→	INVITE(IAM)	
	ACM	+		+	180 Ringing(ACM)	
	ANM	+		+	200 OK INVITE(ANM)	
			Convers	ation		
	FAR	<b>→</b>		<b>→</b>	INFO(FAR)	
				<del>-</del>	200 OK INFO	
	FRJ	+		+	INFO(FRJ)	
				<b>→</b>	200 OK INFO	
	REL	<b>→</b>		<b>→</b>	BYE(REL)	
	RLC	+		<del>-</del>	200 OK BYE(RLC)	

TP414208	SIP reference: RF	C 3261	[4]	Q.191	ISUP reference: 2.5 [1], clauses A.1.1, 2.3 and 4, Q.737 [i.10]		
TSS reference	ISUP-SIP-ISUP/SS/UUS3						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Ensure that a User-to-use confirmed state can succe				n an INFO request during the request is rejected		
SIP parameter	INFO: Content-Type: appl		SUP; FAR co	ntaining the u	ser-to-user indicator		
values	encapsulated in the MIME body INFO: Content-Type: application/ISUP; FRJ containing the user-to-user indicator encapsulated in the MIME body						
ISUP parameter	FAR: User-to-user indicate		e 3 request r	not essential			
values	FRJ: User-to-user indicate						
Comments	SIP-I		SUT		ISUP		
	INVITE(IAM)	<b>→</b>		<b>→</b>	IAM		
	180 Ringing(ACM)	+		+	ACM		
	200 OK INVITE(ANM)	+		+	ANM		
			Conversa				
	INFO(FAR)	→		→	FAR		
	200 OK INFO						
	INFO(FRJ)	+		+	FRJ		
	200 OK INFO	<b>→</b>					
	BYE(REL)	<b>→</b>		<b>→</b>	REL		
	200 OK BYE(RLC)	+		+	RLC		

# Annex A (normative): Test purposes for SIP-I/ISDN interworking

# A.1 Test purposes for ISDN-(ISUP)-SIP-I interworking

# A.1.1 Test purposes for ISDN/SIP Basic call

## A.1.1.1 Interworking from SIP-I to ISDN (Outgoing Call)

### A.1.1.1.1 Sending of the SETUP Message

TP501001	SIP reference: RFC	3261 [4]		Q.	191	ISDN reference: 2.5 [1], clause 6.1.2 (i,1)		
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message							
SIP selection criteria	NOT PICS 4/4 AND NOT PI	ICS 4/5						
ISDN selection criteria	NOT PICS 1/6							
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer including a SDP for A-law (PCMA) and μ-law (PCMU):</li> <li>the SUT shall delete μ-law (PCMU) from the media description that it will send back in the SDP answer;</li> <li>the SUT shall immediately send out the SETUP. The BC is constructed from the ISUP TMR or USI.</li> </ul>							
SIP parameter values	SIP INVITE: Audio RTP/A' 200 OK: Audio RTP/A'							
ISDN parameter values	SETUP BC: A-law							
Comments	SIP-I		SU	Т		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	180 Ringing(ACM)	+			+	ALERTING		
	200 OK INVITE(ANM)	+			+	CONNECT		
	ACK	<b>→</b>						
			Convers	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	+			+	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP501002	SIP reference: RFC	3261 [4	]			ISDN reference:			
		-	-	Q.1	912	2.5 [1], clause 6.1.2 (i,2b			
TSS reference:	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message								
SIP selection criteria	PICS 4/4 AND PICS 4/5								
ISDN selection criteria	NOT PICS 1/6	NOT PICS 1/6							
Test purpose  SIP parameter	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer and 100rel extensions and preconditions extensions in the SIP Supported header: including a SDP for A-law (PCMA) and μ-law (PCMU):</li> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer;</li> <li>the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI.</li> <li>The bearer path shall be connected in both directions when both of the following conditions are satisfied:</li> <li>the I-IWU determines (BCng the procedures defined in RFC 3312 [5]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable).</li> <li>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 [5]) that sufficient preconditions have been met for the session to proceed.</li> <li>SIP INVITE: Audio RTP/AVP 0 8</li> </ul>								
values	200 OK: Audio RTP/AV								
ISDN parameter	SETUP BC: A-law								
values									
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	→							
	183 session Progress	<b>←</b>							
	PRACK	→							
	200 OK PRACK	+							
	UPDATE	<b>→</b>		1	<b>→</b>	SETUP			
	200 OK UPDATE	+							
	180 Ringing(ACM)	+			<del>(</del>	ALERTING			
	200 OK INVITE(ANM)	+		1.	<del>(</del>	CONNECT			
	ACK	<b>→</b>							
			Convers	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+			<del>(</del>	RELEASE			
	` ,				<del>&gt;</del>	RELEASE COMPLETE			

TP501003	SIP reference: RFC 32	261 [4	]			ISDN reference:
				Q.	.191	2.5 [1], clause 6.1.2 (i,1)
TSS reference	SIP-I-ISDN/Basic_call/ Sending	g_of t	he_SETU	P_messa	ge	
SIP selection	NOT PICS 4/4 AND NOT PICS	4/5				
criteria						
ISDN selection	NOT PICS 1/6					
criteria						
Test purpose	Ensure that if the SUT upon re					
	offer including a SDP for A-law					
	·	(PCI	ИU) from t	the media	a des	scription that it will send back in
	the SDP answer;					
	<ul> <li>the SUT shall immediately TMR or USI.</li> </ul>	send	out the S	ETUP. TI	he B	C is constructed from the ISUP
SIP parameter	SIP INVITE: Audio RTP/AVP	8 0				
values	200 OK: Audio RTP/AVP	8				
ISDN parameter	SETUP BC: A-law					
values						
Comments	SIP-I		SU	T		ISDN
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP
	180 Ringing(ACM)	+			+	ALERTING
	200 OK INVITE(ANM)	+			+	CONNECT
	ACK	<b>→</b>				
			Convers	sation		
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT
	200 OK BYE(RLC)	+			<b>←</b>	RELEASE
					<b>→</b>	RELEASE COMPLETE

TP501004	SIP reference: RFC 32	61 [4]		Q.1912	ISDN reference: 2.5 [1], clause 6.1.2 (i,2b)			
TSS reference	SIP-I-ISDN/Basic_call/ Sending	g_of th	ne_SETU	P_message				
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISDN selection criteria	NOT PICS 1/6							
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer and 100rel extensions and preconditions extensions in the SIP Require header including a SDP for A-law (PCMA) and μ-law (PCMU):</li> <li>the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer;</li> <li>the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI.</li> </ul>							
SIP parameter	SIP INVITE: Audio RTP/AVP							
values	200 OK: Audio RTP/AVP	8						
ISDN parameter	SETUP BC: A-law							
values								
Comments	SIP-I		SU	T	ISDN			
	INVITE(IAM)	<b>→</b>						
	183 session Progress	<b>←</b>						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>		<b>→</b>	SETUP			
	200 OK UPDATE	<b>←</b>						
	180 Ringing(ACM)	+		+	ALERTING			
	200 OK INVITE(ANM)	+		+	CONNECT			
	ACK	<b>→</b>	_					
			Convers	sation				
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+		+	RELEASE			
				<b>→</b>	RELEASE COMPLETE			

TP501005	SIP reference: RFC 32	61 [4	]			ISDN reference:	
				Q.	191	2.5 [1], clause 6.1.2 (i,1)	
TSS reference	SIP-I-ISDN/Basic_call/ Sending	g_of tl	he_SETU	P_messa	ge		
SIP selection criteria	NOT PICS 4/4 AND NOT PICS	4/5					
ISDN selection criteria	PICS 1/6						
Test purpose	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer including a SDP for A-law (PCMA) and µ-law (PCMU):  the SUT shall delete A-law (PCMA) from the media description that it will send back in the SDP answer;  the SUT shall immediately send out the SETUP. The BC is constructed from the ISUP TMR or USI.						
SIP parameter	SIP INVITE: Audio RTP/AVP	8 0					
values	200 OK: Audio RTP/AVP	8					
ISDN parameter	SETUP BC: A-law						
values	OID I	T		_		IODNI	
Comments	SIP-I		SU	I		ISDN	
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP	
	180 Ringing(ACM)	+			<b>←</b>	ALERTING	
	200 OK INVITE(ANM)	<b>←</b>			<b>←</b>	CONNECT	
	ACK	<b>→</b>					
			Convers	sation			
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT	
	200 OK BYE(RLC)	<b>←</b>			<del>(</del>	RELEASE	
					<b>→</b>	RELEASE COMPLETE	

TP501006	SIP reference: RF0	3261 [4]				ISDN reference:		
11 301000	On reference. Ki	3 3201 [4]		O.	1912	2.5 [1], clause 6.1.2 (i,2b)		
TSS reference	SIP-I-ISDN/Basic_call/ Sen	ding of the	SETU			[:], ciaco ciii_ (:,=::)		
SIP selection	PICS 4/4 AND PICS 4/5	<u>g_</u> 0o			~gc			
criteria								
ISDN selection	PICS 1/6							
criteria								
Test purpose	Ensure that if the SUT upor	n receipt of	the first	INVITE	with	sufficient digits, with an SDP		
						n the SIP Supported header:		
	including a SDP for <b>A-law</b> (PCMA) and <b>μ-law</b> (PCMU):							
	<ul> <li>the SUT shall delete A</li> </ul>	-law (PCMA	A), if pre	sent, fro	m the	e media description that it will		
	send back in the SDP					·		
	<ul> <li>the SETUP shall be de</li> </ul>	ferred until	all prec	onditions	s hav	e been met. The BC is		
	constructed from the IS							
SIP parameter	SIP INVITE: Audio RTP/A	VP 0 8						
values	200 OK: Audio RTP/A	VP 8						
ISDN parameter	SETUP BC: A-law							
values								
Comments	SIP-I		SU	Т		ISDN		
	INVITE(IAM)	→						
	183 session Progress	<del>-</del>						
	PRACK	→						
	200 OK PRACK	<b>←</b>						
	UPDATE	→			<b>→</b>	SETUP		
	200 OK UPDATE	<b>←</b>						
	180 Ringing(ACM)	<b>←</b>			+	ALERTING		
	200 OK INVITE(ANM)	<b>←</b>			+	CONNECT		
	ACK	<b>→</b>						
		(	Convers	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	+			+	RELEASE		
	200 ON DTL(NLO)	•			~	RELEASE COMPLETE		

TP501007	SIP reference: RFC 32	61 [4	]			ISDN reference:	
						2.5 [1], clause 6.1.2 (i,1)	
TSS reference	SIP-I-ISDN/Basic_call/ Sending	g_of t	he_SETU	P_messa	ge		
SIP selection criteria	NOT PICS 4/4 AND NOT PICS	4/5					
ISDN selection criteria	PICS 1/6						
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer including a SDP for A-law (PCMA) and μ-law (PCMU):</li> <li>the SUT shall delete A-law (PCMA) from the media description that it will send back in the SDP answer;</li> <li>the SUT shall immediately send out the SETUP. The BC is constructed from the ISUP TMR or USI.</li> </ul>						
SIP parameter	SIP INVITE: Audio RTP/AVP	0 8					
values	200 OK: Audio RTP/AVP	8					
ISDN parameter values	SETUP BC: A-law						
Comments	SIP-I		SU	IT		ISDN	
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP	
	180 Ringing(ACM)	+			+	ALERTING	
	200 OK INVITE(ANM)	+			<b>←</b>	CONNECT	
	ACK	<b>→</b>					
			Convers	sation			
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT	
	200 OK BYE(RLC)	<b>←</b>			<b>←</b>	RELEASE	
					<b>→</b>	RELEASE COMPLETE	

TP501008	SIP reference: RFC	3261 [4]				ISDN reference:		
11 001000		0_0.[.]		Q	.1912	2.5 [1], clause 6.1.2 (i,2b)		
TSS reference	SIP-I-ISDN/Basic_call/ Send	ding_of the	SETU					
SIP selection	PICS 4/4 AND PICS 4/5							
criteria								
ISDN selection	PICS 1/6							
criteria								
Test purpose	<ul> <li>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer and 100rel extensions and preconditions extensions in the SIP Require header including a SDP for A-law (PCMA) and μ-law (PCMU):</li> <li>the SUT shall delete A-law (PCMA), if present, from the media description that it will send back in the SDP answer;</li> <li>the SETUP shall be deferred until all preconditions have been met. The BC is constructed from the ISUP TMR or USI. The BC is constructed from the ISUP TMR or USI.</li> </ul>							
SIP parameter	INVITE: Audio RTP/AVP 0 8	3						
values	200 OK: Audio RTP/AVP 8							
ISDN parameter	SETUP: BC: A-law							
values								
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	<b>→</b>						
	183 session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>			<b>→</b>	SETUP		
	200 OK UPDATE	+						
	180 Ringing(ACM)	<b>←</b>			<b>←</b>	ALERTING		
	200 OK INVITE(ANM)	<b>←</b>			<b>←</b>	CONNECT		
	ACK	→						
			Conver	sation				
	BYE(REL)	→			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	+			+	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP501009	SIP reference: RFC	3261 [4	]			ISDN reference:			
		_	_	EN 3	883	001 [2], clause 6.1.3.5.2.2			
TSS reference	SIP-I-ISDN/Basic_call/ Send	ding_of th	ne_SETU	P_messag	ge				
SIP selection	NOT PICS 4/4 AND NOT PI	CS 4/5 A	AND PICS	1/9					
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and								
		A-Law, then independent from the received order of preference.							
	Sends a SETUP message:								
	<ul> <li>the G.711 a-law codec s</li> </ul>	hall be re	eturned in	the SDP	ans	wer as preferred codec.			
SIP parameter	Offer: m=audio 4711 R7	ΓΡ/AVP (	3 C						
values	Answer: m=audio 4712 R	TP/AVP 8	3 0						
ISDN parameter values									
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	180 Ringing(ACM)	+			<del>(</del>	ALERTING			
	200 OK INVITE(ANM)	+			<del>(</del>	CONNECT			
	ACK	<b>→</b>							
			Convers	sation					
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+			<del>(</del>	RELEASE			
					<b>→</b>	RELEASE COMPLETE			

TP501010	SIP reference: RFC 3	3261 [4	]	EN 383	ISDN reference: 001 [2], clause 6.1.3.5.2.2					
TSS reference	SIP-I-ISDN/Basic_call/ Sendir	ng_of tl	ne_SETU	P_message						
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS	1/9							
criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT on receip									
	A-Law and 100rel extensions and preconditions extensions in the SIP Supported header,									
		then independent from the received order of preference:								
	<ul> <li>the SETUP shall be defer</li> </ul>									
	<ul> <li>the G.711 a-law codec sh</li> </ul>	nall be	returned i	n the SDP an	swer as preferred codec.					
SIP parameter	Offer: m=audio 4711 RTF	P/AVP (	8 C							
values	Answer: m=audio 4712 RTF	P/AVP 8	3 0							
ISDN parameter										
values					_					
Comments	SIP-I		SU	IT	ISDN					
	INVITE(IAM)	→								
	183 session Progress	<b>←</b>								
	PRACK	<b>→</b>								
	200 OK PRACK	+								
	UPDATE	<b>→</b>		→	SETUP					
	200 OK UPDATE	+								
	180 Ringing(ACM)	+		+	ALERTING					
	200 OK INVITE(ANM)	+		+	CONNECT					
	ACK	→								
			Conver	sation						
	BYE(REL)	<b>→</b>		→	DISCONNECT					
	200 OK BYE(RLC)	+		+	RELEASE					
				<b>→</b>	RELEASE COMPLETE					

TP501011	SIP reference: RF0	C 3261 [4]		EN 202	ISDN reference:					
T00 (	010 1 10001/0 : 11/ 0	11 6.01	OFTI		001 [2], clause 6.1.3.5.2.2					
TSS reference	SIP-I-ISDN/Basic_call/ Sen			P_message						
SIP selection	PICS 4/4 AND PICS 4/5 AN	ND PICS 1	/9							
criteria										
ISDN selection										
criteria										
Test purpose					a SDP offer for μ-Law and					
	A-Law and 100rel extensions and preconditions extensions in the SIP Require header,									
		then independent from the received order of preference:								
	<ul> <li>the SETUP shall be de</li> </ul>									
	<ul> <li>the G.711 a-law coded</li> </ul>	shall be re	eturned i	n the SDP ar	swer as preferred codec.					
SIP parameter	Offer: m=audio 4711 R	TP/AVP 0	8							
values	Answer: m=audio 4712 R	TP/AVP 8	0							
ISDN parameter										
values										
Comments	SIP-I		SU	IT	ISDN					
	INVITE(IAM)	→								
	183 session Progress	+								
	PRACK	→								
	200 OK PRACK	+								
	UPDATE	<b>→</b>		<b>→</b>	SETUP					
	200 OK UPDATE	+								
	180 Ringing(ACM)	+		+	ALERTING					
	200 OK INVITE(ANM)	+		+	CONNECT					
	ACK	<b>→</b>								
			Convers	sation						
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT					
	200 OK BYE(RLC)	+		+	RELEASE					
	, ,			<b>→</b>	RELEASE COMPLETE					

TP501012	SIP reference: RFC 32	261 [4	]		404	ISDN reference:		
T00 (	010 1 100 1 10 11 11 11 11 11 11 11 11 1		05711			2.5 [1], clause 6.1.2 (i,1)		
TSS reference	SIP-I-ISDN/Basic_call/ Sendin							
SIP selection	NOT PICS 4/4 AND NOT PICS	3 4/5 A	AND PICS	1/9 AND	NO	T PICS 4/19		
criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams and based on operator policy then: if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected.							
SIP parameter	Offer: m=audio 4711 RTP/	AVP	8					
values	m= video 4712 RTP	/AVP	31					
	Answer: m=audio 4711 RTP/ m=video 0 RTP/AVI		8					
ISDN parameter values								
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	180 Ringing(ACM)	+			<del>-</del>	ALERTING		
	200 OK INVITE(ANM)	+			<b>←</b>	CONNECT		
	ACK	<b>→</b>						
			Convers	sation				
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	+			<b>←</b>	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP501013	SIP reference: RFC 3	3261 [4]		Q.1	91:	ISDN reference: 2.5 [1], clause 6.1.2 (i,1)			
TSS reference	SIP-I-ISDN/Basic_call/ Sendi	ng_of th	e_SETUI	P_messag	е				
SIP selection	PICS 4/4 AND PICS 4/5 AND	PICS 1	/9 AND N	NOT PICS	4/1	9			
criteria									
ISDN selection criteria									
Test purpose	one media streams, 100rel exheader: and based on operations of the SDP offer contains one	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams, 100rel extensions and preconditions extensions in the SIP Supported header: and based on operator policy then: if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected.							
SIP parameter	Offer: m=audio 4711 RTF	P/AVP 8							
values	m= video 4712 RT	P/AVP 3	31						
	Answer: m=audio 4711 RTF m=video 0 RTP/AV	-							
ISDN parameter									
values									
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	→							
	183 session Progress	<b>←</b>							
	PRACK	→							
	200 OK PRACK	<b>←</b>							
	UPDATE	<b>→</b>		-	<b>→</b>	SETUP			
	200 OK UPDATE	+							
	180 Ringing(ACM)	+		•	<del>(</del>	ALERTING			
	200 OK INVITE(ANM)	+		•	<del>(</del>	CONNECT			
	ACK	<b>→</b>							
			Convers	sation					
	BYE(REL)	<b>→</b>		[-	<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+		•	<del>(</del>	RELEASE			
				-	<b>→</b>	RELEASE COMPLETE			

TP501014	SIP reference: RFC 3	3261 [4	]	Q.	191	ISDN reference: 2.5 [1], clause 6.1.2 (i,1)		
TSS reference	SIP-I-ISDN/Basic_call/ Sendi	ng_of t	ne_SETU			1		
SIP selection criteria	PICS 4/4 AND PICS 4/5 AND					9		
ISDN selection criteria								
Test purpose	one media streams, 100rel exheader and based on operator if the SDP offer contains one	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams, 100rel extensions and preconditions extensions in the SIP Require needer and based on operator policy then:  If the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected.						
SIP parameter	Offer: m=audio 4711 RTF	P/AVP	3					
values	m= video 4712 RTP/AVP 31							
IODNI .	Answer: m=audio 4711 RTF m=video 0 RTP/A\		3					
ISDN parameter values								
Comments	SIP-I		SU	т		ISDN		
Comments	INVITE(IAM)	→	- 00	' <b>!</b>		IODIN		
	183 session Progress	+						
	PRACK	<b>→</b>						
	200 OK PRACK	+						
	UPDATE	<b>→</b>			<b>→</b>	SETUP		
	200 OK UPDATE	+						
	180 Ringing(ACM)	+			+	ALERTING		
	200 OK INVITE(ANM)	+			+	CONNECT		
	ACK	<b>→</b>						
			Convers	sation				
	BYE(REL)	<b>→</b>			1	DISCONNECT		
	200 OK BYE(RLC)	<b>←</b>			<b>←</b>	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP501015	SIP reference: RFC 3261 [4]			ISDN reference:					
			Q.191	12.5 [1], clause 6.1.2 (i,1)					
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_S	SETU	P_message						
SIP selection		NOT PICS 4/4 AND NOT PICS 4/5 AND PICS 1/9 AND PICS 4/19							
criteria									
ISDN selection criteria									
Test purpose	one media streams and based on open	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams and based on operator policy then: the call is refused with a 415 Unsupported media type response.							
SIP parameter	Offer: m=audio 4711 RTP/AVP 8								
values	m= video 4712 RTP/AVP 31								
ISDN parameter									
values									
Comments	SIP-I		SUT	ISDN					
	INVITE(IAM) →								
	415 Unsupported media type ←								
	ACK →								

TP501016	SIP reference: RFC 3261 [4]	ISDN reference:								
			Q.191	2.5 [1]	], clause 6.1.2 (i,1)					
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message								
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9	9 AND	PICS 4/19							
criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams, 100rel extensions and preconditions extensions in the SIP Supported									
	header: and based on operator polic									
	the call is refused with a 415 Unsuppor	the call is refused with a 415 Unsupported media type response.								
SIP parameter	Offer: m=audio 4711 RTP/AVP 8									
values	m= video 4712 RTP/AVP 31									
ISDN parameter										
values										
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	INVITE(IAM) →								
	415 Unsupported media type	415 Unsupported media type ←								
	ACK -	<b>→</b>								

TP501017	SIP reference: RFC 3261 [4] ISDN reference:									
	-	•	Q.1912	2.5 [1]	], clause 6.1.2 (i,1)					
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message									
SIP selection	PICS 4/4 AND PICS 4/5 AND PICS 1/9 AND PICS 4/19									
criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT on receipt of an	INVIT	E message with	a SDI	offer with more than					
	one media streams, 100rel extension			xtens	ions in the SIP Require					
	header and based on operator poli	cy the	n:							
	the call is refused with a 415 Unsupp	orted r	nedia type respo	nse.						
SIP parameter	Offer: m=audio 4711 RTP/AVP 8									
values	m= video 4712 RTP/AVP 31									
ISDN parameter										
values										
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM) →									
	415 Unsupported media type									
	ACK →									

TP501018	SIP reference: RFC 32	261 [4	]	0		ISDN reference:			
TSS reference	Q.1912.5 [1], clause 6.1.3 SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message								
		g_or t	ne_SETU	P_message	=				
SIP selection	PICS 1/2								
criteria									
ISDN selection	PICS 1/9								
criteria									
Test purpose	Ensure that the SUT on receipt of an INVITE message containing an encapsulated IAM with the media description defined with the "a = " "b =" and "m=" lines set to a_b_m_LINE_VALUE: sends the SETUP message with the Bearer Capability (BC) constructed from the USI parameter in the encapsulated IAM or, if absent, constructed from the TMR of the encapsulated IAM according the ISUP rules.								
SIP parameter values									
ISDN parameter	SETUP; BC Coding standard	: CCI	TT standa	rdized codi	na				
values	Information transfer capabili					or from the TMR			
	transfer mode: circuit mode	.,. o			•	o			
	information transfer rate: 64	khits/	s						
Comments	SIP-I	T T T T T T T T T T T T T T T T T T T	SU	IT		ISDN			
	INVITE(IAM)	<b>→</b>		-	<b>&gt;</b>	SETUP			
	180 Ringing(ACM)	+		4		ALERTING			
	200 OK INVITE(ANM)	+		4	_	CONNECT			
	ACK	<b>→</b>				· · · · · · · · · · · · · · · · · · ·			
			Convers	sation					
	BYE(REL)	<b>→</b>		-	<b>&gt;</b>	DISCONNECT			
	200 OK BYE(RLC)	+		•	-	RELEASE			
				=	<b>&gt;</b>	RELEASE COMPLETE			

	Values for test purposes TP501018 a_b_m_LINE_VALUE								
		m= line b= line a= line		a= line	BC_VALUE Information Transport User Information				
test purpose s	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwidth- value&gt;</bandwidth- </modifier>			User Information Layer 1 Protocol Indicator		
				NOTE: value> for <modifier> of  AS is  evaluated to  be B kbit/s.</modifier>					
VA_01	Audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	Constructed from the encapsulated IAM	"G.711 μ-law"		
VA_02	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> PCMU/8000</dynamic-pt>	Constructed from the encapsulated IAM	"G.711 μ-law"		
VA_03	Audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	Constructed from the encapsulated IAM	"G.711 A-law"		
VA_04	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap: <dynamic-pt> PCMA/8000</dynamic-pt>	Constructed from the encapsulated IAM	"G.711 A-law"		
VA_05	Audio	RTP/AVP	9	AS:64 kbit/s	rTPmap:9 G722/8000	"Unrestricted digital inf. w/tones/ann"			
VA_06	Audio	RTP/AVP	Dynamic PT	AS:64 kbit/s	rtpmap: <dynamic-pt> CLEARMODE/8000</dynamic-pt>	"Unrestricted digital information"			
VA_07	image	Udptl	t38	N/A or up to 64 kbit/s	Based on T.38	Constructed from the encapsulated IAM			
VA_08	image	Tcptl	t38	N/A or up to 64 kbit/s	Based on T.38	Constructed from the encapsulated IAM			

TP501019	SIP reference: RF0	C 3261 [4]	ISDN reference: Q.1912.5 [1], clause 6.1.3.5					
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message							
SIP selection criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message containing an encapsulated IAM, sends an SETUP message with the <b>HLC</b> information element constructed from the encapsulated ATP (HLC).							
SIP parameter values								
ISDN parameter values								
Comments	SIP-I		SU	Т	ISDN			
	INVITE(IAM)	<b>→</b>		-3	SETUP			
	180 Ringing(ACM)	+		€	ALERTING			
	200 OK INVITE(ANM)	+		€	CONNECT			
	ACK	<b>→</b>						
	Conversation							
	BYE(REL)	<b>→</b>		3	DISCONNECT			
	200 OK BYE(RLC)	+		€	RELEASE			
				-	RELEASE COMPLETE			

TP501020	SIP reference: RFC	3261 [4]			ISDN reference:				
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message								
SIP selection									
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT on receipt of an INVITE message with a Called party number +CC NDC SN where CC is the country code of the network in which the next hop terminates, component of the Request-URI send a SETUP message.  Type of number: "National number", remove "+CC" and use the remaining digits to fill the Address signals contained in the user info.  Numbering plan Indicator ISDN (Telephony) numbering plan.								
SIP parameter values									
ISDN parameter values	SETUP : Called party numb	er							
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP				
	180 Ringing(ACM)	+		+	ALERTING				
	200 OK INVITE(ANM)	+		+	CONNECT				
	ACK	<b>→</b>							
			Conversation						
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	+	-	+	RELEASE				
			_	<b>→</b>	RELEASE COMPLETE				

TP501021	SIP reference: RFC 3	261 [4	]		ISDN reference:				
TSS reference	SIP-I-ISDN/Basic_call/ Sending_of the_SETUP_message								
SIP selection									
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT on receipt of an INVITE message with a Called party number +CC NDC SN where CC is the country code of the network in which the next hop terminates, component of the Request-URI send a SETUP message.  Type of number: "Subscriber number", remove "+CC NDC" and use the remaining digits to fill the Address signals contained in the user info.  Numbering plan Indicator ISDN (Telephony) numbering plan.								
SIP parameter values									
ISDN parameter values	SETUP : Called party number	-							
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP				
	180 Ringing(ACM)	+		+	ALERTING				
	200 OK INVITE(ANM)	+		+	CONNECT				
	ACK	<b>→</b>							
			Conversati	on					
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	+		+	RELEASE				
				<b>→</b>	RELEASE COMPLETE				

TP501020	SIP reference: RF0	C 3261 [4]			ISDN reference:			
TSS reference	SIP-I-ISDN/Basic_call/ Ser	nding_of the	_SETUP_me	ssage				
SIP selection								
criteria								
ISDN selection								
criteria								
Test purpose	Ensure that the SUT on receipt of an INVITE message with a Called party number +CC NDC SN where CC is the country code of the network in which the next hop terminates, component of the Request-URI send a SETUP message.  Type of number: "unknown", remove "+CC" and use the remaining digits to fill the Address signals contained in the user info.  Numbering plan Indicator ISDN (Telephony) numbering plan.							
SIP parameter values								
ISDN parameter values	SETUP: Called party numb	er						
Comments	SIP-I		SUT		ISDN			
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP			
	180 Ringing(ACM)	+		+	ALERTING			
	200 OK INVITE(ANM)	+		+	CONNECT			
	ACK	<b>→</b>						
			conversation	1				
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+		+	RELEASE			
				<b>→</b>	RELEASE COMPLETE			

#### A.1.1.1.1 Void

# A.1.1.1.2 Sending of the INFO

TP502001	SIP reference: RFC	3261 [4]			ISDN reference:				
TSS reference	SIP-I-ISDN/Basic_call/Sending_of INFO_message								
SIP selection	PICS 3/4	PICS 3/4							
criteria									
ISDN selection	PICS 3/8								
criteria									
Test purpose  SIP parameter values	<ul> <li>the number of digits already</li> <li>sends a INFO and pass</li> <li>the INFO shall contain</li> </ul>	ereby the or accumuly accumuly accumuly it to outon its Sub	number of digit ated for the ca going ISDN pro sequent Numb	s in the II: cedure: er parai	Request-URI is greater than				
ISDN parameter values									
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP				
	INVITE(SAM)	→		→	INFO				
	INVITE(SAM)	<b>→</b>		<b>→</b>	INFO				
	180 Ringing(ACM)	+		+	ALERTING				
	200 OK INVITE(ANM)	+		+	CONNECT				
	ACK	→							
			Conversation	1					
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	+		+	RELEASE				
				<b>→</b>	RELEASE COMPLETE				

TP502002	SIP reference: RFC 32	61 [4]		ISDN reference:					
TSS reference	SIP-I-ISDN/Basic_call/Sending_of INFO_message								
SIP selection	PICS 3/4								
criteria									
ISDN selection	PICS 3/8								
criteria									
Test purpose	Ensure that the SUT receives an INVITE with the INFOe Call-ID and From tag as a previous INVITE whereby the number of digits in the Request-URI is <b>fewer</b> than the number of digits already accumulated for the call:  • then the SUT shall immediately send a <b>484 Address Incomplete</b> response for this INVITE;  • in this case no INFO is sent to ISDN.								
SIP parameter									
values									
ISDN parameter									
values									
Comments	SIP-I	SU	JT	ISDN					
	INVITE(IAM)	<b>→</b>							
	INVITE(SAM)	<b>→</b>							
	INVITE(SAM)	<b>→</b>							
	484 Address incomplete	+							
	ACK	<b>→</b>							

# A.1.1.1.3 Receipt of the ALERTING - CALL PROCEEDING - PROGRESS message

TP503001	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.5 1)				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message							
SIP selection								
criteria								
ISDN selection criteria	PICS 3/8 AND PICS 1/6	PICS 3/8 AND PICS 1/6						
Test purpose	<ul> <li>the 180 Ringing SIP</li> </ul>	<ul> <li>Ensure that the SUT in call state N25, on receipt the ALERTING message:</li> <li>the 180 Ringing SIP response is sent. Ensure that the in-band information can be transmitted to the calling user.</li> </ul>						
SIP parameter values		The state of the s						
ISDN parameter values								
Comments	SIP-I		SU	IT		ISDN		
	INVITE(IAM)	→			<b>→</b>	SETUP		
	,				<del>-</del>	SETUP ACK		
	180 Ringing(ACM)	+			<del>-</del>	ALERTING		
		band Info						
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	+			<b>←</b>	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP503002	SIP reference: RFC 32	261 [4]		(	<b>ე</b> .19	ISDN reference: 12.5 [1], clause 6.5 1)			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection criteria						-			
ISDN selection criteria									
Test purpose		Ensure that the SUT in call state N6, on receipt the ALERTING message:  a 180 Ringing SIP response is sent. Ensure that the in-band information can be transmitted to the calling user.							
SIP parameter									
values									
ISDN parameter values									
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	180 Ringing(ACM)	+			<b>←</b>	ALERTING			
	Inband	Info							
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+			+	RELEASE			
					<b>→</b>	RELEASE COMPLETE			

TP503003	SIP reference: RFC 32	261 [4]	ISDN reference: Q.1912.5 [1], clause 6.5 2)						
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection									
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT in call sta	te N9,	on receip	t the ALE	RTI	NG message;			
	a 180 Ringing SIP response is sent. Ensure that the in-band information can be transmitted to the calling user.								
SIP parameter									
values									
ISDN parameter									
values									
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			<del>-</del>	CALL PROCEEDING			
	180 Ringing(CPG)	+			<del>-</del>	ALERTING			
	Inband	Info							
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+			<b>←</b>	RELEASE			
			-		<b>→</b>	RELEASE COMPLETE			

TP503004	SIP reference: RFC 32	261 [4]		ISDN reference:		
				(	Q.19	12.5 [1], clause 6.5 2)
TSS reference	SIP-I-ISDN/Basic_call/Receipt	of AL	ERTING_	_CALL-PF	ROC	_PROGRESS_message
SIP selection						
criteria						
ISDN selection	PICS 3/8 AND PICS 1/6					
criteria						
Test purpose	Ensure that the SUT in call sta	te N25	5, on rece	ipt of the	CAL	L PROCEEDING message:
	<ul> <li>a 183 Session Progress w</li> </ul>	ith an	encapsul	ated ACM	1 is s	sent to the previous entity.
SIP parameter	183 Session Progress encapsu	ulated	ACM: BC	i Called p	arty	status = no indication
values						
ISDN parameter	CALL PROCEEDING					
values						
Comments	SIP-I		SU	IT		ISDN
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP
	183 Session Progress(ACM)	+			<del>-</del>	CALL PROCEEDING
	CANCEL(REL)	<b>→</b>			<b>→</b>	RELEASE
	200 OK CANCEL(RLC)	+			+	RELEASE COMPLETE

TP503005	SIP reference: RFC 32	61 [4			0.40	ISDN reference:	
						912.5 [1], clause 6.5 2)	
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	_of AL	<u>.ERTING</u>	_CALL-PI	ROC	C_PROGRESS_message	
SIP selection							
criteria							
ISDN selection							
criteria							
Test purpose	Ensure that the SUT in call state N6, on receipt of the CALL PROCEEDING message containing a <b>progress indicator</b> set to PI_VALUE:  a 183 Session Progress with an encapsulated ACM is sent to the previous entity.						
SIP parameter	183 Session Progress encapsu	ılated	ACM: BC	i Called p	arty	status = no indication, ATP	
values	with Progress indicator			•	•		
ISDN parameter	CALL PROCEEDING						
values							
Comments	SIP-I		SU	IT		ISDN	
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP	
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROCEEDING(PI)	
	CANCEL(REL)	<b>→</b>		•	<b>→</b>	RELEASE	
	200 OK CANCEL(RLC)	+			<del>-</del>	RELEASE COMPLETE	

TP503006	SIP reference: RFC 32	61 [4]	]		Q.19	ISDN reference: 012.5 [1], clause 6.5 2)				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message									
SIP selection	·					•				
criteria										
ISDN selection										
criteria										
Test purpose	progress indicator set to PI_\	Ensure that the SUT in call state N9, on receipt of the PROGRESS message containing a <b>progress indicator</b> set to PI_VALUE:								
CID managed an						is sent to the previous entity.				
SIP parameter values	183 Session Progress with end									
values	183 Session Progress with end Progress indicator	apsui	aled CPG	• Evenum	uica	ioi= Progress, ATP with				
ISDN parameter values	CALL PROCEEDING									
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	<b>←</b>			<b>←</b>	CALL PROCEEDING				
	183 Session Progress(CPG)	<b>←</b>			+	PROGRESS(PI)				
	CANCEL(REL)	<b>→</b>			<b>→</b>	RELEASE				
	200 OK CANCEL(RLC)	+			+	RELEASE COMPLETE				

TP503007	SIP reference: RF	C 3261 [4]		Q.19	ISDN reference: 912.5 [1], clause 6.5 2)					
TSS reference	SIP-I-ISDN/Basic_call/Rec	SIP-I-ISDN/Basic call/Receipt of ALERTING CALL-PROC PROGRESS message								
SIP selection criteria										
ISDN selection criteria	PICS 1/6	PICS 1/6								
Test purpose	progress indicator set to	Ensure that the SUT in call state N9, on receipt of the ALERTING message containing a <b>progress indicator</b> set to PI_VALUE: the 180 Ringing SIP response is sent.								
SIP parameter	180 Ringing encapsulated	ACM: BCi c	alled party s	tatus=su	bscriber free, ATP with					
values	Progress indicator									
ISDN parameter values	ALERTING(PI)									
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP					
	180 Ringing(ACM)	+		+	ALERTING(PI)					
				•						
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT					
	200 OK BYE(RLC)	+		+	RELEASE					
				<b>→</b>	RELEASE COMPLETE					

TP503008	SIP reference: RFC 3261 [4]			Q.	ISDN reference: 1912.5 [1], clause 6.5 2)				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection criteria									
ISDN selection criteria	PICS 1/6	PICS 1/6							
Test purpose	progress indicator set to	Ensure that the SUT in call state N25, on receipt of a ALERTING message containing the progress indicator set to PI_VALUE: the 180 Ringing SIP response is sent.							
SIP parameter values	180 Ringing encapsulated Progress indicator	180 Ringing encapsulated ACM: BCi called party status=subscriber free, ATP with							
ISDN parameter values	ALERTING(PI)								
Comments	SIP-I		SU'	Т	ISDN				
	INVITE(IAM)	<b>→</b>		→	SETUP				
	INVITE(SAM)	<b>→</b>		<b>→</b>	INFO				
	180 Ringing(ACM)	180 Ringing(ACM) ← ALERTING(PI)							
	BYE(REL)	<b>→</b>		-	DISCONNECT				
	200 OK BYE(RLC)	<del>-</del>		<del>-</del>	2.00020.				
				<b>→</b>					

TP503009	SIP reference: RFC 32	261 [4]	ISDN reference: Q.1912.5 [1], clause 6.5 2)						
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection criteria						-			
ISDN selection criteria	PICS 1/6								
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message in state N6, where the end of address signalling is determined by analysis of the called party number to indicate that <b>a sufficient number of digits has been received</b> to route the call to the called party, on receipt of a CALL PROCEEDING: a 183 Session Progress with an encapsulated ACM is sent to the previous entity.								
SIP parameter values									
ISDN parameter values									
Comments	SIP-I		SL	JT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+		-	+	CALL PROCEEDING			
	CANCEL(REL)	<b>→</b>			<b>→</b>	RELEASE			
	200 OK CANCEL(RLC)	+		•	+	RELEASE COMPLETE			

TP503010	SIP reference: RFC 32	261 [4]		ISDN reference:					
				Q.19	912.5 [1], clause 6.5 2)				
TSS reference	SIP-I-ISDN/Basic_call/Receipt	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message							
SIP selection									
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in call state N25, on receipt of a PROGRESS message containing no progress indicator:  a 183 Session Progress with an encapsulated ACM is sent to the previous entity.								
SIP parameter	183 Session Progress encaps		_						
values	103 Session Flogress encaps	ulateu ACI	vi. bCi	Called party	status = 110 iriulcation				
ISDN parameter	CALL PROCEEDING								
values									
Comments	SIP-I		SU	Т	ISDN				
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP				
	183 Session Progress(ACM)	<b>←</b>		+	CALL PROCEEDING				
				+	PROGRESS				
	CANCEL(REL)	<b>→</b>		<b>→</b>	RELEASE				
	200 OK CANCEL(RLC)	+		+	RELEASE COMPLETE				

TP503011	SIP reference: R	FC 3261 [4]		Q.	ISDN reference: 1912.5 [1], clause 6.6				
TSS reference	SIP-I-ISDN/Basic_call/Re	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message							
SIP selection criteria									
ISDN selection criteria	PICS 1/6	PICS 1/6							
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, receives an ALERTING message, having sent a 180 Ringing message, on receipt of a PROGRESS message:  • the PROGRESS is not interworked.								
SIP parameter values									
ISDN parameter values									
Comments	SIP-I		SUT	Γ	ISDN				
	INVITE(IAM)	→		→	SETUP				
	180 Ringing(ACM)	+		+	ALERTING				
				+	PROGRESS				
	BYE(REL)	→	•	<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	+	•	+	RELEASE				
				→	RELEASE COMPLETE				

TP503012	SIP reference: RFC 32	261 [4]	]		ISDN reference:			
<i>(</i>	Q.1912.5 [1], clause 6.6							
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	_of AL	<u>.ERTING</u>	_CALL-PI	ROC	_PROGRESS_message		
SIP selection								
criteria								
ISDN selection	PICS 1/6							
criteria								
Test purpose		ě						
SIP parameter values	183 Session Progress: Encaps	ulated	d ACM, ca	alled party	/ sta	tus indicator=no indication		
ISDN parameter values								
Comments	SIP-I		SU	IT		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROCEEDING		
					<b>←</b>	PROGRESS		
	CANCEL(REL)	<b>→</b>			<b>→</b>	RELEASE		
	200 OK CANCEL(RLC)	+		•	<b>←</b>	RELEASE COMPLETE		

TP503013	SIP reference: RFC 32	61 [4]		ISDN reference:				
					Q.1912.5 [1], clause 6.6			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of AL	ERTING.	_CALL-PF	ROC	_PROGRESS_message		
SIP selection								
criteria								
ISDN selection	PICS 1/6							
criteria								
Test purpose						E message sends out a SETUP		
	message, receives a CALL PR					es an ALERTING:		
	<ul> <li>sends a 180 Ringing with encapsulated CPG Alerting.</li> </ul>							
SIP parameter	183 Session Progress with end					status indicator=no indication		
values	180 Ringing encapsulated CPC	3: Eve	nt indicat	or=Alertin	ıg			
ISDN parameter								
values								
Comments	SIP-I		SU	JT		ISDN		
	INVITE(IAM)	<b>←</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM)	+			+	CALL PROCEEDING		
	180 Ringing(CPG)							
	Inband	Inband Info						
	BYE(REL) → →				<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	+			+	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

TP503014	SIP reference: RFC 32	261 [4]			ISDN reference:			
					1912.5 [1], clause 6.6			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message							
SIP selection								
criteria								
ISDN selection	PICS 1/6							
criteria								
Test purpose					E message sends out a SETUP			
	message, receives a CALL PR							
	with a progress indicator who		progress	s description	value is set to PI_VALUE, on			
	receipt of a ALERTING Messa	•						
	sent a 180 Ringing message.							
SIP parameter	183 Session Progress with encapsulated ACM: called party status indicator=no indication							
values	183 Session Progress with end	capsula	ited CPG	event indica	ator=Progress, ATP with			
	Progress indicator							
1001	180 Ringing encapsulated CPC	Evel :ز	nt indicat	or=Alerting				
ISDN parameter values								
Comments	SIP-I		SU	IT	ISDN			
	INVITE(IAM)	→		→	SETUP			
	183 Session Progress(ACM)	<b>←</b>		+	CALL PROCEEDING			
	183 Session Progress(CPG)			<b>←</b>	PROGRESS(PI)			
	180 Ringing(CPG)	<b>←</b>		<b>←</b>	ALERTING			
	Inband	Info						
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+	•	+	RELEASE			
				<b>→</b>	RELEASE COMPLETE			

TP503015	SIP reference: RFC 32	261 [4]			Q.1	ISDN reference: 912.5 [1], clause 6.6			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message								
SIP selection criteria									
ISDN selection criteria	PICS 1/6								
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, on receipt of a ALERTING Message, receives a PROGRESS message with a <b>progress indicator</b> where the progress description value is set to PI_VALUE:  • sent a 183 Session Progress message containing a encapsulated CPG.								
SIP parameter	180 Ringing encapsulated ACM								
values	183 Session Progress with end Progress indicator	apsul	ated CPG	6: event ind	dica	tor=Progress, ATP with			
ISDN parameter values									
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	180 Ringing(ACM)	+			<del>(</del>	ALERTING			
	183 Session Progress(CPG) ← PROGRESS(PI)								
	Inband Info								
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+	·		<del>(</del>	RELEASE			
					<b>→</b>	RELEASE COMPLETE			

TP503016	SIP reference: RFC 3	261 [4]			Q.	ISDN reference: 1912.5 [1], clause 6.6		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of ALERTING_CALL-PROC_PROGRESS_message							
SIP selection criteria								
ISDN selection criteria	PICS 1/6	PICS 1/6						
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, receives a CALL PROCEEDING, receives a PROGRESS message with a progress indicator where the progress description value is set to PI_VALUE:  • no message is sent.							
SIP parameter values	183 Session Progress with en- 183 Session Progress with en- Progress indicator					status indicator=no indication tor=Progress, ATP with		
ISDN parameter values								
Comments	SIP-I		SU	Т		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM)	+			+	CALL PROCEEDING		
	183 Session Progress(CPG)				+	PROGRESS(PI)		
	Inband Info							
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT		
	200 OK BYE(RLC)	+			<b>←</b>	RELEASE		
					<b>→</b>	RELEASE COMPLETE		

# A.1.1.1.4 Receipt of the CONNECT Message

TP504001	SIP reference: RFC 326	1 [4]		Q.1	ISDN reference: 1912.5 [1], clause 6.7	
TSS reference	SIP-I-ISDN/Basic_call/Receipt_c	of CONNEC	T_message			
SIP selection	·					
criteria						
ISDN selection criteria						
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, receives an ALERTING message, on receipt of a CONNECT message:  • sends a 200 OK INVITE to the previous entity.  The bearer path shall be connected in both directions when the following condition is satisfied:  • the BICC outgoing bearer set-up procedure, (see ITU-T Recommendation Q.1902.4 [Error! Reference source not found.]) is successfully completed.  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 [5]) that sufficient preconditions have been met for the session to proceed.					
SIP parameter values						
ISDN parameter						
values						
Comments	SIP-I		SUT		ISDN	
	INVITE(IAM)	→		<b>→</b>	SETUP	
	180 Ringing(ACM)	<b>←</b>		+	ALERTING	
		<b>←</b>		<b>←</b>	CONNECT	
	ACK	→				
		Conv	ersation			
	BYE(REL)	→		<b>→</b>	DISCONNECT	
	200 OK BYE(RLC)	<del>(</del>		<b>←</b>	RELEASE	
				<b>→</b>	RELEASE COMPLETE	

TP504002	SIP reference: RFC	3261 [4	]		ISDN reference:
					1912.5 [1], clause 6.7
TSS reference	SIP-I-ISDN/Basic_call/Rece	ipt_of C	ONNECT_	_message	
SIP selection					
criteria					
ISDN selection					
criteria					
Test purpose					message sends out a SETUP
	message, receives a ALERT	I ING me	ssage, or	receipt of a	CONNECT message
OID 1	sends a 200 OK INVITE to t	ne previ	ous entity		
SIP parameter values					
ISDN parameter					
values					
Comments	SIP-I		SU	JT	ISDN
	INVITE(IAM)	→			
	183 session Progress	+			
	PRACK	<b>→</b>			
	200 OK PRACK	+			
	UPDATE	<b>→</b>		→	SETUP
	200 OK UPDATE	+			
	180 Ringing(ACM)	+		+	ALERTING
	200 OK INVITE(ANM)	+		+	CONNECT
	ACK	<b>→</b>			
			Conver	sation	
	BYE(REL)	<b>→</b>		→	DISCONNECT
	200 OK BYE(RLC)	+		+	RELEASE
				→	RELEASE COMPLETE

TP504003	SIP reference: RFC 326	1 [4]				ISDN reference:	
					<b>Q</b> .1	1912.5 [1], clause 6.7	
TSS reference	SIP-I-ISDN/Basic_call/Receipt_c	f CO	NNECT_r	nessage			
SIP selection criteria							
ISDN selection criteria							
Test purpose	SDP offer was not received in the initial INVITE. Ensure that the SUT, having received the ALERTING message, on receipt of an CONNECT message:  • sends a 200 OK INVITE to the UAC. The 200 OK INVITE shall include an SDP offer consistent with the BC used.						
SIP parameter	200 OK SDP offer						
values	ACK SDP answer						
ISDN parameter values							
Comments	SIP-I		SU	Т		ISDN	
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP	
	180 Ringing(ACM)	+			<del>(</del>	ALERTING	
	200 OK INVITE(ANM; SDP1)	+			<del>(</del>	CONNECT	
	ACK(SDP2)	<b>→</b>					
	Conversation						
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT	
	200 OK BYE(RLC)	+			<del>(</del>	RELEASE	
					<b>→</b>	RELEASE COMPLETE	

TP504004	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.4, 6.7			
TSS reference	SIP-I-ISDN/Basic_call/Recei	pt_of CO	ONNECT_r	nessage			
SIP selection							
criteria							
ISDN selection							
criteria							
Test purpose	<ul> <li>SDP offer was received in the initial INVITE. Ensure that the SUT, on receipt of an CON message:</li> <li>sends a 200 OK INVITE to the previous entity.</li> </ul>						
SIP parameter		200 OK INVITE: encapsulated CON					
values							
ISDN parameter							
values							
Comments	SIP-I		SUT	•		ISDN	
	INVITE(IAM)	<b>→</b>		-	• (	SETUP	
	200 OK INVITE(CON)	+		•	1	CONNECT	
	ACK	<b>→</b>					
	Conversation						
	BYE(REL)	<b>→</b>		-	<b>→</b> [	DISCONNECT	
	200 OK BYE(RLC)	+		•	<b>.</b>	RELEASE	
				-	<b>→</b>	RELEASE COMPLETE	

### A.1.1.1.5 Initiation of the release procedure from the ISDN side

TP505001	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2							
TSS reference	SIP-I-ISDN/Basic_call/Rece	ipt_of_DIS0	C_or_RELE	ASE				
SIP selection	NOT PICS 4/10	-						
criteria								
ISDN selection								
criteria								
SIP parameter values	SETUP message, on receip CV_ISDN, location LOC_IS  the SUT immediately the SUT shall send to	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, on receipt of an RELEASE COMPLETE message with the Cause value CV_ISDN, location LOC_ISDN:  the SUT immediately requests the disconnection of the internal bearer path;  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.  SIP Statue-Code: SIP_FAILURE_VA (PIXIT)						
ISDN parameter	REL_COMP: cause value:	CV_ISDN (	PIXIT)					
values								
Comments	SIP-I		SUT		ISDN			
	INVITE(IAM)	<b>→</b>	•	<b>→</b>	SETUP			
	SIP_FAILURE_VA(REL)	<b>←</b>	•	+	RELEASE COMPLETE			
	ACK 🗦							

TP505002	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2						
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE						
SIP selection criteria	NOT PICS 4/10		_				
ISDN selection criteria							
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, on receipt of a RELEASE with the <b>Cause value</b> CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN REL_COMP is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.						
SIP parameter values	SIP Statue-Code: SIP_FAILURE_V						
ISDN parameter values	RELEASE; cause value: CV_ISD	N (PIXIT)					
Comments	SIP-I	SU	JT		ISDN		
	INVITE(IAM)	<b>)</b>		<b>→</b>	SETUP		
	SIP_FAILURE_VA(REL)	-		<del>(</del>	RELEASE		
	ACK → RELEASE COMPLETE						

TP505003	SIP reference: RFC 3261 [4] ISDN reference:								
	Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of_DI	SC_or_R	ELEASE					
SIP selection criteria	NOT PICS 4/10	NOT PICS 4/10							
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, on receipt of an DISCONNECT message with the <b>Cause value</b> CV_ISDN, location LOC_ISDN:  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN REL_COMP is returned to the ISDN side;  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.								
SIP parameter	SIP Statue-Code: SIP_FAILURE	_VA (P	'IXII)						
values	DICC:OV ICDN	/DIVIT	٠,						
ISDN parameter values	DISC; cause value: CV_ISDN	(PIXII	)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	SIP_FAILURE_VA(REL)	+			<del>(</del>	DISCONNECT			
	ACK	<b>→</b>			<b>→</b>	RELEASE			
					<del>(</del>	RELEASE COMPLETE			

TP505004	SIP reference: RFC 32	ე 19	ISDN reference: 12.5 [1], clause 6.11.2					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection criteria	NOT PICS 4/10							
ISDN selection criteria								
Test purpose  SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, and on receipt of a RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  the SUT immediately requests the disconnection of the internal bearer path;  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.							
values	SIP Statue-Code: SIP_FAILURE	_v^ (i	IXII)					
ISDN parameter values	REL_COMP: cause value: CV	_ISDI	N (PIXIT)					
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	<b>^</b>			<b>→</b>	SETUP		
					+	SETUP ACK		
	SIP_FAILURE_VA(REL)	+			+	RELEASE COMPLETE		
	ACK	<b>→</b>						

TP505005	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TCC veference									
TSS reference	•	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose  SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, and on receipt of a RELEASE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, a REL_COMP is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.								
values	SIP Statue-Code: SIP_FAILURE	_vA (	PIAII)						
ISDN parameter values	RELEASE; cause value: CV_I	SDN	(PIXIT)						
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
					<b>←</b>	SETUP ACK			
	SIP_FAILURE_VA(REL)	<b>←</b>			<b>←</b>	RELEASE			
	ACK → RELEASE COMPLETE								

TP505006	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	NOT PICS 4/10								
criteria									
ISDN selection									
criteria									
SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, and on receipt of a DISCONNECT message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side;  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.  SIP Statue-Code: SIP_FAILURE_VA (PIXIT								
ISDN parameter	DISC: cause value: CV_ISDN	(PIXI	T)						
values		(· // ·	- /						
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
		← SETUP ACK							
	SIP_FAILURE_VA(REL) ← DISCONNECT								
	ACK	<b>→</b>				RELEASE			
					<b>←</b>	RELEASE COMPLETE			

TP505007	SIP reference: RFC 32	(	ISDN reference: Q.1912.5 [1], clause 6.11.2					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection criteria	NOT PICS 4/10							
ISDN selection criteria								
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, on receipt of a re-INVITE message sends an INFORMATION message and on receipt of a REL EASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN  the SUT immediately requests the disconnection of the internal bearer path.  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA							
SIP parameter values	SIP Statue-Code: SIP_FAILURE	_VA (	PIXII)					
ISDN parameter values	REL_COMP: cause value: CV_	_ISDI	N (PIXIT)					
Comments	SIP-I		SU	T		ISDN		
	INVITE(IAM)	1			<b>↑</b>	SETUP		
					+	SETUP ACK		
	INVITE(IAM)	1			<b>↑</b>	INFO		
	SIP_FAILURE_VA(REL)	+			+	RELEASE COMPLETE		
	ACK	1						

TP505008	SIP reference: RFC 3261 [4] ISDN reference:							
						12.5 [1], clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection								
criteria								
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, on receipt of a re-INVITE message sends an INFORMATION message and on receipt of a REL EASE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.							
SIP parameter	SIP Statue-Code: SIP_FAILURE	_VA (F	PIXIT)					
values ISDN parameter values	RELEASE; cause value: CV_IS	SDN (	PIXIT)					
Comments	SIP-I		SU	IT		ISDN		
	INVITE(IAM)	<b>→</b>	•		1	SETUP		
					<b>+</b>	SETUP ACK		
	INVITE(IAM)	<b>→</b>			1	INFO		
	SIP_FAILURE_VA(REL)	+			+	RELEASE		
	ACK	<b>→</b>			<b>^</b>	RELEASE COMPLETE		

TP505009	SIP reference: RFC 32		Q.19	ISDN reference: 912.5 [1], clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of_DISC_or_	RELEASE			
SIP selection criteria	NOT PICS 4/10					
ISDN selection criteria						
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a SETUP ACKNOWLEDGE message, on receipt of a re-INVITE message sends an INFORMATION message and on receipt of a DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN:  the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side;  the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.					
SIP parameter values	SIP Statue-Code: SIP_FAILURE	_VA (PIXIT				
ISDN parameter values	DISC: cause value: CV_ISDN	(PIXIT)				
Comments	SIP-I	0)	SUT		ISDN	
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP	
				+	SETUP ACK	
	INVITE(IAM)	<b>→</b>		<b>→</b>	INFO	
	SIP_FAILURE_VA(REL)	+		+	DISCONNECT	
	ACK	<b>→</b>		<b>→</b>	RELEASE	
				+	RELEASE COMPLETE	

	Values for test purp	oses TP108001 - TP108009
	←SIP Message	← REL
	SIP_FAILURE_VA	Cause Indicators parameter CV_ISDN,
VA_1	404 Not Found	Cause Value No. 1 ("unallocated (unassigned) number")
VA_2	500 Server Internal Error	Cause Value No. 2 ("no route to network")
VA_3	500 Server Internal Error	Cause Value No. 3 ("no route to destination")
VA_4	500 Server Internal Error	Cause Value No. 4 ("Send special information tone")
VA_5	404 Not Found	Cause Value No. 5 ("Misdialled trunk prefix")
VA_6	500 Server Internal Error	Cause Value No. 8 ("Pre-emption")
VA_7	500 Server Internal Error	Cause Value No. 9 ("Pre-emption-circuit reserved for reuse")
VA_8	486 Busy Here	Cause Value No. 17 ("user busy")
VA_9	480 Temporarily unavailable	Cause Value No. 18 ("no user responding")
VA_10	480 Temporarily unavailable	Cause Value No. 19 ("no answer from the user")
VA_11	480 Temporarily unavailable	Cause Value No. 20 ("subscriber absent")
VA_12	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")
VA_13	410 Gone	Cause Value No. 22 ("number changed")
VA_14	480 Temporarily unavailable	Cause Value No. 25 ("Exchange routing error")
VA_15	502 Bad Gateway	Cause Value No. 27 ("destination out of order")
VA_16	484 Address Incomplete	Cause Value No. 28 ("invalid number format (address incomplete")
VA_17	500 Server Internal Error	Cause Value No. 29 ("facility rejected")
VA_18	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified") (Class default)
VA_19	486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible)	Cause Value in the Class 010 (No circuit/channel available, Cause Value No. 34)
	else 480 Temporarily unavailable	
VA_20	500 Server Internal Error	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 ("requested facility not subscribed")
VA_22	500 Server Internal Error (SIP-I only)	Cause Value No. 55 ("incoming calls barred within CUG")
VA_23 VA_24	500 Server Internal Error 500 Server Internal Error	Cause Value No. 57 ("bearer capability not authorized")  Cause Value No. 58 ("bearer capability not presently")
VA_24 VA_25	500 Server Internal Error	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_26	500 Server Internal Error	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 to 79) (79 is class default)
VA_27	500 Server Internal Error	Cause Value No. 87 ("user not member of CUG")
VA_27 VA_28	500 Server Internal Error	Cause Value No. 88 ("incompatible destination")
VA_20 VA 29	500 Server Internal Error	Cause Value No. 90 ("Non-existent CUG")
VA_29 VA_30	404 Not Found	Cause Value No. 91 ("invalid transit network selection")
VA_31	500 Server Internal Error	Cause Value No. 95 ("invalid message") (Class default)
VA_32	500 Server Internal Error	Cause Value No. 97 ("Message type non-existent or not implemented")
VA_33	500 Server Internal Error	Cause Value No. 99 ("information element/parameter non- existent or not implemented")
VA_34	480 Temporarily unavailable	Cause Value No. 102 ("recovery on timer expiry")
VA_35	500 Server Internal Error	Cause Value No. 103 ("Parameter non-existent or not implemented, pass on")
VA_36	500 Server Internal Error	Cause Value No. 110 ("Message with unrecognized Parameter, discarded")
VA_37	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)
VA_38	480 Temporarily unavailable	Cause Value No. 127 ("interworking unspecified") (Class default)

TP505010	SIP reference: RFC 32	61 [4]			Q.19	ISDN reference: 12.5 [1], clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection criteria	NOT PICS 4/10							
ISDN selection criteria								
Test purpose  SIP parameter values ISDN parameter	Ensure that the SUT in the Idle message, and receives a CALL COMPLETE message with the  the SUT immediately request the SUT shall send the apple 183 Session Progress encapsus SIP Statue-Code: SIP_FAILURE REL_COMP: cause value: CV	PROC Cause ests the propria lated A _VA (P	CEEDING  e value Ce e disconr te SIP st ACM: BC	G message V_ISDN nection of atus defined the street of the	ge, c , <b>loc</b> f the ned	on receipt of a RELEASE  ation LOC_ISDN: internal bearer path; as SIP_FAILURE_VA.		
values	0.0	1 1		_	1	liani		
Comments	SIP-I	<b>→</b>	SU	I	<b>→</b>	ISDN SETUD		
	INVITE(IAM) 183 Session Progress(ACM)	7 +			7	SETUP CALL PROC		
	SIP_FAILURE_VA(REL)	+			<del>-</del>	RELEASE COMPLETE		
	ACK	<b>→</b>						

TP505011	SIP reference: RFC 3261 [4] ISDN reference:							
				(	Q.19	12.5 [1], clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection								
criteria								
Test purpose  SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a RELEASE message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.							
values	183 Session Progress encapsu SIP Statue-Code: SIP_FAILURE			i Calleu p	Jaily	status = 110 indication		
ISDN parameter values	RELEASE; cause value: CV_I							
Comments	SIP-I		SU	IT		ISDN		
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP		
	183 Session Progress(ACM)	+			+	CALL PROC		
	SIP_FAILURE_VA(REL)	+		•	+	RELEASE		
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE		

TP505012	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	NOT PICS 4/10								
criteria									
ISDN selection									
criteria									
SIP parameter values	<ul> <li>Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN:         <ul> <li>the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side;</li> <li>the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.</li> </ul> </li> <li>183 Session Progress encapsulated ACM: BCi Called party status = no indication SIP Statue-Code: SIP_FAILURE_VA (PIXIT)</li> </ul>								
values	DISC; cause value: CV_ISDN	(FIXI	1)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	183 Session Progress(ACM) ← CALL PROC							
	SIP_FAILURE_VA(REL) ← DISCONNECT								
	ACK	→				RELEASE			
					<b>←</b>	RELEASE COMPLETE			

TP505013	SIP reference: RFC 32	61 [4	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2							
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection	NOT PICS 4/10									
criteria										
ISDN selection criteria										
Test purpose  SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a progress indicator PI_VALUE, on receipt of a RELEASE COMPLETE message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.  183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator									
ISDN parameter	SIP Statue-Code: SIP_FAILUR REL_COMP: cause value: CV									
values	Cause value. Ov	_1001	<b>v</b> (1 1/11)							
Comments	SIP-I		SU	ΙΤ		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	+			+	CALL PROC				
	183 Session Progress(CPG)	<b>←</b>			<b>←</b>	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	<b>←</b>			<b>←</b>	RELEASE COMPLETE				
	ACK	<b>→</b>								

TP505014	SIP reference: RFC 32	61 [4	]			ISDN reference:			
	Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	NOT PICS 4/10								
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT in the Idle								
	message, receives a CALL PRO	OCE	EDING me	essage fo	llow	ed by a PROGRESS message			
	with a progress indicator PI_\			eipt of a F	KELE	EASE message with the Cause			
	value CV_ISDN, location LOC	_			£ 41	Section of the second section (Allege Alege			
						internal bearer path. When the			
	bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message								
	is returned to the ISDN sid	,	oto CID of	tatua dafi	d	oo CID TAILLIDE VA			
SIP parameter	<ul> <li>the SUT shall send the app</li> <li>183 Session Progress encapsu</li> </ul>								
values	183 Session Progress with enc								
values	Progress indicator	apsui	aleu Oi C	LVEIILIII	luica	tor= 1 logiess, All with			
	SIP Statue-Code: SIP_FAILURE	VΔ (	DIXIT)						
ISDN parameter	RELEASE; cause value: CV_I								
values	release, sauce value: ev_i	00.1	(1.17.11)						
Comments	SIP-I		SU	JT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	183 Session Progress(CPG) ← PROGRESS(PI)								
	SIP_FAILURE_VA(REL)	<b>←</b>			+	RELEASE			
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE			

TP505015	SIP reference: RFC 32	261 [4	]	(	<b>ე</b> .19	ISDN reference: 12.5 [1], clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a progress indicator PI_VALUE, on receipt of a DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.								
SIP parameter values	183 Session Progress encapsu 183 Session Progress with enc Progress indicator SIP Statue-Code: SIP_FAILURE	apsu	ated CPG						
ISDN parameter values	DISC; cause value: CV_ISDN								
Comments	SIP-I		SU	IT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	<b>←</b>			+	CALL PROC			
	183 Session Progress(CPG)	+			<b>←</b>	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	<b>←</b>			+	DISCONNECT			
	ACK	→			<b>→</b>	RELEASE			
					<b>←</b>	RELEASE COMPLETE			

Table 21

	purpose TP1080010- TP1080015	
	←SIP Message SIP_FAILURE_VA	← REL  Cause Indicators parameter  CV_ISDN,
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_2	480 Temporarily unavailable	Cause Value No. 18 ("No user responding")
VA_3	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")
VA_4	410 Gone	Cause Value No. 22 ("number changed")
VA_5	502 Bad Gateway	Cause Value No. 27 ("destination out of order")
VA_6	484 Address Incomplete	Cause Value No. 28 ("invalid number format (address incomplete")
VA_7	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified") (Class default)
VA_8	500 Server Internal Error	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 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)

TP505016	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message sends out a SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE COMPLETE message:  • the SUT immediately requests the disconnection of the internal bearer path;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.								
SIP parameter values	SIP Statue-Code: SIP_FAILURE								
ISDN parameter values	REL_COMP: cause value: CV	_ISDI	N (PIXIT)						
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
					<b>←</b>	SETUP ACK			
	180 Ringing(ACM)	+			+	ALERTING			
	SIP_FAILURE_VA(REL)	+		•	+	RELEASE COMPLETE			
	ACK	<b>→</b>							

TP505017	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	NOT PICS 4/10									
ISDN selection criteria										
SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE message:  • the SUT immediately requests the disconnection of the internal bearer path When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.  SIP Statue-Code: SIP_FAILURE_VA (PIXIT)									
ISDN parameter	RELEASE; cause value: CV_IS	SDN	(PIXIT)							
values			,							
Comments	SIP-I		SU	IT		ISDN				
	INVITE(IAM)	1			1	SETUP				
					<b>+</b>	SETUP ACK				
	180 Ringing(ACM)	180 Ringing(ACM) ←   ←   ALERTING								
	SIP_FAILURE_VA(REL)	SIP_FAILURE_VA(REL) ← RELEASE								
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE				

TP505018	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose  SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN DISCONNECT message:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.  SIP Statue-Code: SIP_FAILURE_VA (PIXIT								
values									
ISDN parameter values	DISC: cause value: CV_ISD	N (PIXI	T)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>^</b>	SETUP			
					<b>←</b>	SETUP ACK			
	180 Ringing(ACM)	180 Ringing(ACM) ← ← ALERTING							
	SIP_FAILURE_VA(REL)	+		_	+	DISCONNECT			
	ACK	<b>→</b>			<b>→</b>	RELEASE			
					+	RELEASE COMPLETE			

TP505019	SIP reference: RFC 32	61 [4	(	ISDN reference: Q.1912.5 [1], clause 6.11.2					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE COMPLETE message:  • the SUT immediately requests the disconnection of the internal bearer path;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.								
SIP parameter values	183 Session Progress encapsu SIP Statue-Code: SIP_FAILURE			i Called	oarty	status = no indication			
ISDN parameter values	REL_COMP: cause value: CV	_ISDI	N (PIXIT)						
Comments	SIP-I		SU	ΙΤ		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	180 Ringing(CPG)	+			+	ALERTING			
	SIP_FAILURE_VA(REL)	+		·	+	RELEASE COMPLETE			
	ACK	<b>→</b>							

TP505020	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE message:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side.  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.								
SIP parameter values	183 Session Progress encaps			i Called p	arty	status = no indication			
ISDN parameter values	RELEASE; cause value: CV_								
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	180 Ringing(CPG)	+			+	ALERTING			
	SIP_FAILURE_VA(REL)	+			+	RELEASE			
	ACK	<b>→</b>			<b>^</b>	RELEASE COMPLETE			

TP505021	SIP reference: RFC 32	:61 [4]		O 40	ISDN reference:			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of DISC or			12.5 [1], clause 6.11.2			
SIP selection criteria	NOT PICS 4/10							
ISDN selection criteria								
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN DISCONNECT message:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.							
SIP parameter	183 Session Progress encapsu		Ci Called p	oarty	status = no indication			
values	SIP Statue-Code: SIP_FAILURE							
ISDN parameter values	DISC; cause value: CV_ISDN	(PIXIT)						
Comments	SIP-I	9	UT		ISDN			
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP			
	183 Session Progress(ACM)	+		<b>←</b>	CALL PROC			
	180 Ringing(CPG) ← ALERTING							
	SIP_FAILURE_VA(REL)	+		+	DISCONNECT			
	ACK	→		<b>→</b>	RELEASE			
				+	RELEASE COMPLETE			

TP505022	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose  SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, receives a PROGRESS message with a <b>progress indicator</b> where the progress description value is set to PI_VALUE, on receipt of a ALERTING Message, having sent a 180 Ringing message on receipt of an RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.  183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with								
	Progress indicator SIP Statue-Code: SIP_FAILUR	F V/	(PIXIT)						
ISDN parameter values	REL_COMP: cause value: CV								
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	183 Session Progress(CPG)	+				PROGRESS(PI)			
	180 Ringing(CPG)	+			<b>←</b>	ALERTING			
	SIP_FAILURE_VA(REL)	<b>←</b>			<b>←</b>	RELEASE COMPLETE			
	ACK →								

TP505023	SIP reference: RFC 32	61 [4]	]	(	Q.19	ISDN reference: 112.5 [1], clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, receives a PROGRESS message with a <b>progress indicator</b> where the progress description value is set to PI_VALUE, on receipt of a ALERTING Message, having sent a 180 Ringing message on receipt of an RELEASE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.								
SIP parameter values	183 Session Progress encapsu 183 Session Progress with enc Progress indicator SIP Statue-Code: SIP_FAILUR	apsul	ated CPG						
ISDN parameter values	RELEASE; cause value: CV_I	SDN	(PIXIT)						
Comments	SIP-I		SU	JT		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	183 Session Progress(CPG)	+			<b>←</b>	PROGRESS(PI)			
	180 Ringing(CPG)	+			<b>←</b>	ALERTING			
	SIP_FAILURE_VA(REL)	+			<b>←</b>	RELEASE			
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE			

TP505024	SIP reference: RFC 32	61 [4	]			ISDN reference:		
						12.5 [1], clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	of_D	ISC_or_R	ELEASE				
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection								
criteria								
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP							
	message, receives a CALL PROCEEDING message, receives a PROGRESS message							
	with a progress indicator whe							
	receipt of a ALERTING Messag							
	DISCONNECT message with the							
	• the SUT immediately requests the disconnection of the internal bearer path. When the							
	bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message							
	is returned to the ISDN sid	- ,	-4- OID -4	- 4 l - <b>f</b> '		OID FAILLIDE MA		
OID	the SUT shall send the app							
SIP parameter	183 Session Progress encapsu							
values	183 Session Progress with enc	apsui	ated CPG	Event in	aica	tor= Progress, ATP with		
	Progress indicator SIP Statue-Code: SIP FAILURE	\/A (	DIVIT\					
ISDN parameter	DISC; cause value: CV_ISDN	_ ,						
values	DISC, cause value. CV_ISDN	(FIXI	1)					
Comments	SIP-I		SU	Т		ISDN		
	INVITE(IAM)	<b>→</b>		-	<b>→</b>	SETUP		
	183 Session Progress(ACM)	<del>-</del>			<del>-</del>	CALL PROC		
	183 Session Progress(CPG)	+			+	PROGRESS(PI)		
	180 Ringing(CPG)	+			<del>-</del>	ALERTING		
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT		
	ACK	<b>→</b>			<b>→</b>	RELEASE		
					<b>←</b>	RELEASE COMPLETE		

TP505025	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	NOT PICS 4/10									
ISDN selection criteria										
Test purpose  SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, on receipt of a ALERTING Message, having sent a 180 Ringing message, receives a PROGRESS message with a <b>progress indicator</b> where the progress description value is set to PI_VALUE, on receipt of an RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.  183 Session Progress encapsulated ACM: BCi Called party status = no indication									
Values	183 Session Progress with end Progress indicator SIP Statue-Code: SIP_FAILURE	•		Event in	uica	ioi= Flogress, ATF with				
ISDN parameter values	REL_COMP: cause value: CV									
Comments	SIP-I		SU <sup>*</sup>	Т		ISDN				
	INVITE(IAM)	<b>→</b>		-	1	SETUP				
	183 Session Progress(ACM)	+			4	CALL PROC				
	180 Ringing(CPG)	+	•	·	4	ALERTING				
	183 Session Progress(CPG)	+			+	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	+			+	RELEASE COMPLETE				
	ACK	<b>→</b>	•	·						

TP505026	SIP reference: RFC 32	IP reference: RFC 3261 [4] ISDN reference:								
	Q.1912.5 [1], clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection	NOT PICS 4/10									
criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, on receipt of a ALERTING message, having sent a 180 Ringing message, receives a PROGRESS message with a <b>progress indicator</b> where the progress description value is set to PI_VALUE, on receipt of an RELEASE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.									
SIP parameter	183 Session Progress encapsu									
values	183 Session Progress with enc	apsul	ated CPG	Event in	dica	tor= Progress, ATP with				
	Progress indicator	\/A /	DIVIT\							
ICDM noromotor	SIP Statue-Code: SIP_FAILURE									
ISDN parameter values	RELEASE; cause value: CV_IS	SDIN	(PIXII)							
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM) ← CALL PROC									
	180 Ringing(CPG)	<b>←</b>			<b>←</b>	ALERTING				
	183 Session Progress(CPG)	+			<b>←</b>	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	+				RELEASE				
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE				

TP505027	SIP reference: RFC 32	61 [4]		C	Չ.19	ISDN reference: 0.1912.5 [1], clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message sends out a SETUP message, receives a CALL PROCEEDING message, on receipt of a ALERTING Message, having sent a 180 Ringing message, receives a PROGRESS message with a <b>progress indicator</b> where the progress description value is set to PI_VALUE, on receipt of an DISCONNECT message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA.  183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with								
	Progress indicator SIP Statue-Code: SIP_FAILURE	VA (PI)	(IT)						
ISDN parameter values	DISC; cause value: CV_ISDN		,						
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROC			
	180 Ringing(CPG) ← ALERTING								
	183 Session Progress(CPG) ← PROGRESS(PI)								
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT			
	ACK	→			<b>→</b>	RELEASE			
					<b>←</b>	RELEASE COMPLETE			

Table 22

	Values for test purposes TP108016 and TP108027						
←SIP Message SIP_FAILURE_VA		← REL  Cause Indicators parameter  CV_ISDN,					
VA_1	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")					
VA_2	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified") (Class default)					
VA_4	500 Server Internal Error	Cause Value No. 38 ("Network out of order")					
VA_4	500 Server Internal Error	Cause Value No. 41 ("Temporary failure ")					
VA_5	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)					

TP505028	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	NOT PICS 4/10	NOT PICS 4/10								
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  the SUT immediately requests the disconnection of the internal bearer path;  the SUT shall send a BYE message.									
SIP parameter values	183 Session Progress encaps			i Called p	arty	status = no indication				
ISDN parameter values	REL_COMP: cause value: CV	/_ISD	N (PIXIT)							
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	+		·	+	CALL PROC				
	200 OK INVITE(ANM) ← CONNECT									
			Commun	ication						
	BYE(REL)	+			<b>←</b>	RELEASE COMPLETE				
	200 OK BYE(RLC)	<b>→</b>								

TP505029	SIP reference: RFC 3261 [4] ISDN reference:								
TSS reference	Q.1912.5 [1], clause 6.11.2  SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose  SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;  • the SUT shall send a BYE message.  183 Session Progress encapsulated ACM: BCi Called party status = no indication								
values ISDN parameter	RELEASE; cause value: CV_I	ISDNI (	DIYIT)						
values	TELEAGE, Cause value. CV_I	ISDIN (	1 1/11)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM) ← CALL PROC								
	200 OK INVITE(ANM) ← CONNECT								
			Commun	ication					
	BYE(REL)	+			<b>←</b>	RELEASE			
	200 OK BYE(RLC)	<b>→</b>			<b>→</b>	RELEASE COMPLETE			

TP505030	SIP reference: RFC 32	261 [4	.]		ე 19	ISDN reference: 12.5 [1], clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	NOT PICS 4/10								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side;  • the SUT shall send a BYE message.								
SIP parameter values	183 Session Progress encapsu			i Called p	oarty	status = no indication			
ISDN parameter	DISC; cause value: CV_ISDN	(PIXI	T)						
values									
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM) ← CALL PROC								
	200 OK INVITE(ANM) ← CONNECT								
	Communication								
	BYE(REL)	+			<b>←</b>	DISCONNECT			
	200 OK BYE(RLC)	<b>→</b>			<b>→</b>	RELEASE			
				•	+	RELEASE COMPLETE			

TP505031	SIP reference: RFC 3261 [4] ISDN reference:									
	Q.1912.5 [1], clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection	NOT PICS 4/10									
criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sending out a SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path;  • the SUT shall send a BYE message.									
SIP parameter values										
ISDN parameter values	REL_COMP: cause value: CV	_ISD	N (PIXIT)							
Comments	SIP-I		SU	T		ISDN				
	INVITE(IAM) → SETUP									
	200 OK INVÍTE(ANM) ← CONNECT									
			Commun	ication						
	BYE(REL)	+			<b>+</b>	RELEASE COMPLETE				
	200 OK BYE(RLC)	<b>→</b>								

TP505032	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	NOT PICS 4/10									
ISDN selection criteria										
Test purpose  SIP parameter values	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;  • the SUT shall send a BYE message.									
ISDN parameter	RELEASE; cause value: CV_IS	DN (I	DIXIT)							
values	TELLINOL, GAUGE VAIGE. OV_IC	ا) ۱۱ (۱	17311)							
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP					
	200 OK INVITE(ANM) ← ← CONNECT									
		(	Communication							
	BYE(REL)	<b>←</b>		+	RELEASE					
	200 OK BYE(RLC)	<b>→</b>		<b>→</b>	RELEASE COMPLETE					

TP505033	SIP reference: RFC 32	61 [4	]			ISDN reference:			
	Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	NOT PICS 4/10								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out an SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side;  • the SUT shall send a BYE message.								
SIP parameter values									
ISDN parameter values	DISC; cause value: CV_ISDN	(PIXI	T)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	200 OK INVITE(ANM) ← CONNECT								
	Communication								
	BYE(REL)	+			+	DISCONNECT			
	200 OK BYE(RLC)	<b>→</b>		-	<b>↑</b>	RELEASE			
					<b>+</b>	RELEASE COMPLETE			

TP505034	SIP reference: RFC 3261 [4]			reference: I], clause 6.11.2						
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	NOT PICS 4/10	NOT PICS 4/10								
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on SETUP message, via a broadcast data  the SUT shall send a 480 Tempora	link,	after time-out of 1	Г303	:					
SIP parameter values	480 Temporarily unavailable: Encapsula	ated	REL with cause 1	8						
ISDN parameter values										
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	INVITE(IAM) → SETUP								
	→ SETUP									
	T303 expiry									
	480 Temporarily unavailable(REL)	<b>←</b>								
	ACK	<b>→</b>	•							

#### Table 23

	Values for test purpose TP108029 and TP 108035							
	←SIP Message SIP_FAILURE_VA	← REL  Cause Indicators parameter  CV_ISDN,						
VA_1	BYE	Cause Value No. 16						
VA_2	BYE	Cause Value No. 31 ("normal unspecified") (Class default)						
VA_3	BYE	Cause Value No. 38 ("Network out of order")						
VA_4	BYE	Cause Value No. 41 ("Temporary failure ")						
VA_5	BYE	Cause Value No. 111 ("protocol error, unspecified") (Class default)						

TP505035	SIP reference: RFC 3261 [4] ISDN reference:									
	Q.1912.5 [1], clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_o	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	PICS 4/10	PICS 4/10								
ISDN selection criteria										
Test purpose  SIP parameter	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends out a SETUP message, on receipt of an ISDN RELEASE COMPLETE, where the cause value defined as CV_ISDN:  • 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 ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.  SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)									
values ISDN parameter	REL COMP: cause value: CV I	SDN (DIYIT)								
values	Cause value. Cv_i	ODIN (FIXIT)								
Comments	SIP-I	SU	IT	ISDN						
	INVITE(IAM)	<b>&gt;</b>	→	SETUP						
	SIP_FAILURE_VA(REL)	E	<del>(</del>	RELEASE COMPLETE						
	ACK -	<b>&gt;</b>								

TP505036	SIP reference: RFC 3261	[4]	Q.1	ISDN reference: Q.1912.5 [1], clause 6.11.2					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sending out a SETUP message, on receipt of an ISDN REL, where the cause value defined as CV_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.								
SIP parameter values	SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)								
ISDN parameter values	RELEASE; cause value: CV_ISDN (PIXIT)								
Comments	SIP-I SUT ISDN								
		<b>&gt;</b>	<b>→</b>	SETUP					
	SIP_FAILURE_VA(REL)	<b>-</b>	<b>+</b>	RELEASE					
	ACK -	<b>→</b>	→	RELEASE COMPLETE					

Table 24

	Values for test purposes TP108036, TP108037						
	←SIP Message SIP_FAILURE_VA CV SIP	← REL  Cause Indicators parameter  CV ISDN					
VA_1	404 Not Found Cause Value No. 1	Cause Value No. 1 ("unallocated (unassigned) number")					
VA_2	500 Server Internal Error Cause Value No. 2	Cause Value No. 2 ("no route to network")					
VA_3	500 Server Internal Error Cause Value No. 3	Cause Value No. 3 ("no route to destination")					
VA_4	500 Server Internal Error Cause Value No. 4	Cause Value No. 4 ("Send special information tone")					
VA_5	404 Not Found Cause Value No. 5	Cause Value No. 5 ("Misdialled trunk prefix")					
VA_6	500 Server Internal Error Cause Value No. 8	Cause Value No. 8 ("Preemption")					
VA_7	500 Server Internal Error Cause Value No. 9	Cause Value No. 9 ("Preemption-circuit reserved for reuse")					
VA_8	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")					
VA_9	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("no user responding")					
VA_10	480 Temporarily unavailable Cause Value No. 19	Cause Value No. 19 ("no answer from the user")					
VA_11	480 Temporarily unavailable Cause Value No. 20	Cause Value No. 20 ("subscriber absent")					
VA_12	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")					
VA_13	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")					
VA_14	480 Temporarily unavailable Cause Value No. 25	Cause Value No. 25 ("Exchange routing error")					
VA_15	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")					
VA_16	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete")					
VA_17	500 Server Internal Error	Cause Value No. 29 ("facility rejected")					

		poses TP108036, TP108037					
	←SIP Message	← REL					
	SIP_FAILURE_VA CV_SIP	Cause Indicators parameter CV_ISDN					
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 to 79	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 to 79) (79 is class default)					
VA_27	500 Server Internal Error Cause Value No. 87	Cause Value No. 87 ("user not member of CUG")					
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)					

TP505037	SIP reference: RFC 32	61 [4]		ISDN reference: Q.1912.5 [1], clause 6.11.2						
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	PICS 4/10									
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a RELEASE COMPLETE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  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 ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.									
SIP parameter	183 Session Progress encapsu									
values	SIP Statue-Code: SIP_FAILURE			eason hea	ade	r value: CV_SIP (PIXIT)				
ISDN parameter values	REL_COMP: cause value: CV_ISDN (PIXIT)									
Comments	SIP-I SUT ISDN									
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	+			<del>(</del>	CALL PROC				
	SIP_FAILURE_VA(REL)	+			<del>(</del>	RELEASE COMPLETE				
	ACK	<b>→</b>								

TP505038	SIP reference: RFC 3261 [4] ISDN reference:									
	Q.1912.5 [1], clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection	PICS 4/10									
criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a RELEASE message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side:									
	<ul><li>the SUT shall send the app</li><li>the ISDN Cause Value field</li></ul>									
	header field.									
SIP parameter values	183 Session Progress encapsu SIP Statue-Code: SIP_FAILURE									
ISDN parameter values	RELEASE; cause value: CV_ISDN (PIXIT)									
Comments	SIP-I		SU	T T		ISDN				
	INVITE(IAM)	<b>→</b>			1	SETUP				
	183 Session Progress(ACM)	+			+	CALL PROC				
	SIP_FAILURE_VA(REL)	+			4	RELEASE				
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE				

TP505039	SIP reference: RFC 3261 [4] ISDN reference:									
T00 (	Q.1912.5 [1], clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	_ot_DI	SC_or_R	ELEASE						
SIP selection	PICS 4/10									
criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, and receives a CALL PROCEEDING message, on receipt of a DISCONNECT message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.									
SIP parameter	183 Session Progress encapsu									
values	SIP Statue-Code: SIP_FAILURE			eason ne	ade	r value: CV_SIP (PIXIT)				
ISDN parameter	DISC; cause value: CV_ISDN	(PIXI	1)							
values						1				
Comments	SIP-I		SU	<u>IT</u>		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	<b>←</b>			<b>←</b>	CALL PROC				
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT				
	ACK	<b>→</b>			<b>→</b>	RELEASE				
					<b>←</b>	RELEASE COMPLETE				

TP505040	SIP reference: RFC 32	61 [4	]			ISDN reference:				
						012.5 [1], clause 6.11.2				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection	PICS 4/10									
criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a <b>progress indicator</b> PI_VALUE, on receipt of a RELEASE COMPLETE message									
	with the Cause value CV_ISDN					3				
	<ul> <li>the SUT immediately reque</li> </ul>	ests tl	ne discon	nection o	f the	internal bearer path;				
	<ul> <li>the SUT shall send the app</li> </ul>									
	<ul> <li>the ISDN Cause Value field header field.</li> </ul>	d in th	ne ISDN F	REL mess	age	is mapped to the Reason				
SIP parameter	183 Session Progress encapsu									
values	183 Session Progress with enc	apsul	ated CPG	Event in	dica	tor= Progress, ATP with				
	Progress indicator									
	SIP Statue-Code: SIP_FAILURE			eason he	ade	r value: CV_SIP (PIXIT)				
ISDN parameter values	REL_COMP: cause value: CV	_ISDI	N (PIXIT)							
Comments	SIP-I		SU	IT		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	+			<b>←</b>	CALL PROC				
	183 Session Progress(CPG)	<b>←</b>			<b>←</b>	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	+			+	RELEASE COMPLETE				
	ACK	<b>→</b>		_						

TP505041	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2									
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	PICS 4/10									
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a progress indicator PI_VALUE, on receipt of a RELEASE message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE COMPLETE message is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.									
SIP parameter values	183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator SIP Statue-Code: SIP_FAILURE_VA (PIXIT), Reason header value: CV_SIP (PIXIT)									
ISDN parameter values	RELEASE; cause value: CV_ISDN (PIXIT)									
Comments	SIP-I		SU	JT		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	+			+	CALL PROC				
	183 Session Progress(CPG)	+			+	PROGRESS(PI)				
	SIP_FAILURE_VA(REL)	<b>←</b>			<b>←</b>	RELEASE				
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE				

TP505042	SIP reference: RFC 32		ISDN reference: Q.1912.5 [1], clause 6.11.2						
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	PICS 4/10								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message with a progress indicator PI_VALUE, on receipt of a DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE message is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.								
SIP parameter values	183 Session Progress encapsu 183 Session Progress with enc Progress indicator SIP Statue-Code: SIP_FAILURE	apsul	ated CPG	Event in	dica	tor= Progress, ATP with			
ISDN parameter values	DISC; cause value: CV_ISDN			243011110	auc	value. ov_on (FDAT)			
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	<b>←</b>			<del>(</del>	CALL PROC			
	183 Session Progress(CPG)	+			<del>(</del>	PROGRESS(PI)			
	SIP_FAILURE_VA(REL)	+			+	DISCONNECT			
	ACK	<b>→</b>			<b>→</b>	RELEASE			
					<del>(</del>	RELEASE COMPLETE			

Table 25

	Values for test purpose TP108038 - TP108043						
	←SIP Message SIP_FAILURE_VA CV_SIP	← REL Cause Indicators parameter CV_ISDN,					
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)					

TP505043	SIP reference: RFC 32	rence: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE									
SIP selection criteria	PICS 4/10									
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends a SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE COMPLETE message:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.									
SIP parameter values	SIP Statue-Code: SIP_FAILURE	_VA (PIXIT),	Reason he	eade	r value: CV_SIP (PIXIT)					
ISDN parameter values	REL_COMP: cause value: CV_ISDN (PIXIT)									
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP					
				<b>←</b>	SETUP ACK					
	180 Ringing(ACM)	+		<b>←</b>	ALERTING					
	SIP_FAILURE_VA(REL)	+		<b>←</b>	RELEASE COMPLETE					
	ACK	<b>→</b>								

TP505044	SIP reference: RFC 32	61 [4]			O 10	ISDN reference: 12.5 [1], clause 6.11.2		
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE							
SIP selection criteria	PICS 4/10							
ISDN selection criteria								
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends a SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN RELEASE message:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.							
SIP parameter values	SIP Statue-Code: SIP_FAILURE	_VA (I	PIXIT), <b>R</b> e	eason he	ade	r value: CV_SIP (PIXIT)		
ISDN parameter values	RELEASE; cause value: CV_IS	SDN (	PIXIT)					
Comments	SIP-I		SU	IT		ISDN		
	INVITE(IAM)	<b>→</b>	•	•	<b>→</b>	SETUP		
					+	SETUP ACK		
	180 Ringing(ACM)	+			<b>←</b>	ALERTING		
	SIP_FAILURE_VA(REL)	<b>←</b>			<b>←</b>	RELEASE		
	ACK	<b>→</b>			<b>→</b>	RELEASE COMPLETE		

TP505045	SIP reference: RFC 32	61 [4]			ISDN reference:				
					12.5 [1], clause 6.11.2				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	PICS 4/10								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message sends a SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, having sent a 180 Ringing message on receipt of an ISDN DISCONNECT message:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE is returned to the ISDN side;  • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.								
SIP parameter	SIP Statue-Code: SIP_FAILURE_	VA (PIXIT	), Reason he	eade	r value: CV_SIP (PIXIT)				
values		•	-						
ISDN parameter values	DISC: cause value: CV_ISDN (	(PIXIT)							
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP				
				+	SETUP ACK				
	180 Ringing(ACM)	+		+	ALERTING				
	SIP_FAILURE_VA(REL)	+		+	DISCONNECT				
	ACK	<b>→</b>		<b>→</b>	RELEASE				
				+	RELEASE COMPLETE				

Table 26

	Values for test purposes TP108044 and TP108046						
	←SIP Message SIP_FAILURE_VA CV_SIP	← REL Cause Indicators parameter CV_ISDN,					
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)					

TP505046	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE COMPLETE message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path;  • the SUT shall send a BYE message;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.								
SIP parameter	183 Session Progress encaps								
values	SIP Statue-Code: SIP_FAILURE			eason he	ade	r value: CV_SIP (PIXIT)			
ISDN parameter values	REL_COMP: cause value: CV	_ISDI	N (PIXIT)						
Comments	SIP-I		SU	T		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	200 OK INVITE(ANM)	+			+	CONNECT			
			Commun	ication					
	BYE(REL)	+			4	RELEASE COMPLETE			
	200 OK BYE(RLC)	<b>→</b>							

TP505047	SIP reference: RFC 32	:61 [4	]	(	Q.19	ISDN reference: 12.5 [1], clause 6.11.2			
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	PICS 4/10								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side;  • the SUT shall send a BYE message;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.								
SIP parameter values	183 Session Progress encapsu								
ISDN parameter	SIP Statue-Code: SIP_FAILURE RELEASE: cause value: CV_I			ason ne	aue	Value. CV_SIF (FIXIT)			
values	NELEAGE. Cause value. CV_	ODIN	(11/11)						
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	200 OK INVITE(ANM)	+			+	CONNECT			
			Commun	ication					
	BYE(REL)	+			+	RELEASE			
	200 OK BYE(RLC)	<b>→</b>			<b>→</b>	RELEASE COMPLETE			

TP505048	SIP reference: RFC 32	261 [4	]		<b>1</b> 0	ISDN reference:			
TSS reference	Q.1912.5 [1], clause 6.11.2   SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	PICS 4/10	_01_D	19C_01_k	ELEASE					
criteria	PICS 4/10								
ISDN selection									
criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CALL PROCEEDING message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path When the bearer channel is available for re-selection, an ISDN RELEASE is returned to the ISDN side;  • the SUT shall send a BYE message;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.								
SIP parameter values	183 Session Progress encapsu SIP Statue-Code: SIP_FAILURE								
ISDN parameter values	DISC: cause value: CV_ISDN			Judon 116	auc	· • • • • • • • • • • • • • • • • • • •			
Comments	SIP-I		SU	ΙT		ISDN			
	INVITE(IAM)	<b>→</b>			1	SETUP			
	183 Session Progress(ACM)	+			<b>+</b>	CALL PROC			
	200 OK INVITE(ANM)	+			+	CONNECT			
			Commun	ication					
	BYE(REL)	+			+	DISCONNECT			
	200 OK BYE(RLC)	<b>→</b>			<b>→</b>	RELEASE			
					<b>←</b>	RELEASE COMPLETE			

TP505049	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection criteria	PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE COMPLETE message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. the SUT shall send a BYE message;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.								
SIP parameter values	SIP Statue-Code: SIP_FAILURE	_VA (	(PIXIT), <b>Re</b>	ason head	ler value: CV_SIP (PIXIT)				
ISDN parameter values	REL_COMP: cause value: CV	_ISD	N (PIXIT)						
Comments	SIP-I		SU	T	ISDN				
	INVITE(IAM)	1		7	SETUP				
	200 OK INVITE(ANM)	+		+	CONNECT				
			Commun	ication					
	BYE(REL)	+		+	RELEASE COMPLETE				
	200 OK BYE(RLC)	<b>→</b>							

TP505050	SIP reference: RFC 3261 [4] ISDN reference: Q.1912.5 [1], clause 6.11.2								
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_DISC_or_RELEASE								
SIP selection	PICS 4/10								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a RELEASE message with the Cause value CV_ISDN, location LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RLC is returned to the ISDN side;  • the SUT shall send a BYE message;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.								
SIP parameter values	SIP Statue-Code: SIP_FAILURE_	VA (F	PIXIT), <b>Rea</b>	ison hea	de	r value: CV_SIP (PIXIT)			
ISDN parameter values	RELEASE: cause value: CV_IS	DN (	PIXIT)						
Comments	SIP-I		SUT	'		ISDN			
	INVITE(IAM)	<b>→</b>		-	<b>→</b>	SETUP			
	200 OK INVITE(ANM)	<del>(</del>		•	<b>←</b>	CONNECT			
			Communic	ation					
	BYE(REL)	<b>←</b>		•	<del>(</del>	RELEASE			
	200 OK BYE(RLC)	<b>→</b>		-	<b>→</b>	RELEASE COMPLETE			

TP505051	SIP reference: RFC 3	3261 [4	1]	1	Q.19	ISDN reference: 112.5 [1], clause 6.11.2	
TSS reference	SIP-I-ISDN/Basic_call/Receip	t_of_D	ISC_or_R	ELEASE			
SIP selection criteria	PICS 4/10						
ISDN selection criteria							
Test purpose	Ensure that the SUT in the Idle state on receipt of an INVITE message, sends a SETUP message, receives a CONNECT message, a 200 OK message is sent, on receipt of a DISCONNECT message with the <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN:  • the SUT immediately requests the disconnection of the internal bearer path. When the bearer channel is available for re-selection, an ISDN RELEASE is returned to the ISDN side;  • the SUT shall send a BYE message;  • the ISDN Cause Value field in the ISDN REL message is mapped to the Reason header field.						
SIP parameter values	SIP Statue-Code: SIP_FAILUR	RE_VA (	(PIXIT), <b>R</b> e	eason he	eade	r value: CV_SIP (PIXIT)	
ISDN parameter values	DISC: cause value: CV_ISDI	N (PIXI	T)				
Comments	SIP-I		SU	ΙT		ISDN	
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP	
	200 OK INVITE(ANM)	+			<b>←</b>	CONNECT	
			Commun	ication			
	BYE(REL)	+			+	DISCONNECT	
	200 OK BYE(RLC)	<b>→</b>			<b>→</b>	RELEASE	
					<b>←</b>	RELEASE COMPLETE	

Table 27

	Values for test purposes TP108047 and TP108052						
←SIP Message SIP_FAILURE_VA CV_SIP		← REL Cause Indicators parameter CV_ISDN,					
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)					

# A.1.1.1.6 Receipt of BYE / CANCEL messages

TP506001	SIP reference: RF0	C 3261 [4]		ISDN reference: Q.1912.5 [1], clause 6.11.1						
TSS reference	SIP-I-ISDN/Basic_call/Rece	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection criteria										
ISDN selection criteria										
Test purpose	SETUP message, receives <b>BYE</b> , the SUT shall send a	Ensure that the SUT in the Idle state on receipt of an INVITE message, sending out a SETUP message, receives an ALERTING and CONNECT message. On receipt of SIP BYE, the SUT shall send an ISDN DISCONNECT with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side.								
SIP parameter values										
ISDN parameter values	DISC: Cause value and loc	ation mappe	ed from the e	encapsu	lated REL in the received BYE					
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP					
	180 Ringing(ACM)	+		+	ALERTING					
	200 OK INVITE(ANM)	<del>(</del>		<b>←</b>	CONNECT					
	ACK	<b>→</b>								
			onversatio	n						
	BYE(REL)	<b>→</b>		<b>→</b>	DISCONNECT					
	200 OK BYE(RLC)	+		+	RELEASE					

TP506002	SIP reference: RFC 3	261 [4]		ISDN reference: Q.1912.5 [1], clause 6.11.1						
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL									
SIP selection										
criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out a SETUP message, receives an ALERTING message, the SUT on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side.									
SIP parameter values										
ISDN parameter	DISC: Cause value and location	on map	ped from the	encapsul	ated REL in the received					
values	CANCEL									
Comments	SIP-I		SUT		ISDN					
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP					
	180 Ringing(ACM)	<b>←</b>		<b>←</b>	ALERTING					
	CANCEL(REL)	<b>→</b>		<b>→</b>	DISCONNECT					
	200 OK CANCEL	+		+	RELEASE					
	487 Request Terminated	+		<b>→</b>	RELEASE COMPLETE					
	ACK	<b>→</b>								

TP506003	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, the SUT on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISCONNECT with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side.								
SIP parameter values									
ISDN parameter values	DISC: Cause value and location	on map	ped from	the enca	psul	ated REL in the received			
Comments	SIP-I		SU	ΙΤ		ISDN			
	INVITE(IAM)	<b>→</b>							
	100 Trying	+			<b>→</b>	SETUP			
	CANCEL(REL)	<b>→</b>			<b>→</b>	DISCONNECT			
	200 OK CANCEL	+			<del>(</del>	RELEASE			
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE			
	ACK	<b>→</b>							

TP506004	SIP reference: RFC 3261 [4]				ISDN reference: Q.1912.5 [1], clause 6.11.1				
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a SETUP ACKNOWLEDGE message, the SUT on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISCONNECT with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side.								
SIP parameter values									
ISDN parameter values	DISC: Cause values and loca CANCEL	ition ma	pped fron	n the er	capsi	ulated REL in the received			
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	, ,				+	SETUP ACK			
	CANCEL(REL) → DISCONNECT								
	200 OK CANCEL	+			+	RELEASE			
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE			
	ACK	<b>→</b>							

TP506005	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.11.1						
TSS reference	SIP-I-ISDN/Basic_call/Receip	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection										
criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a SETUP ACKNOWLEDGE message, and on receipt of a INFO message on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side.									
SIP parameter										
values										
ISDN parameter	DISC: Cause value and locat	ion mappe	ed from	the encapsu	lated REL in the received					
values	CANCEL									
Comments	SIP-I		SU	T	ISDN					
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP					
				<b>←</b>	SETUP ACK					
	INVITE(IAM)	<b>→</b>		→	INFO					
	CANCEL(REL)	<b>→</b>		→	DISCONNECT					
	200 OK CANCEL	+	•	+	RELEASE					
	487 Request Terminated	+		→	RELEASE COMPLETE					
	ACK	<b>→</b>	•							

TP506006	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection									
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, on receipt of SIP CANCEL, the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side.								
SIP parameter values	183 Session Progress encapsu	ulated	ACM: BC	i Called	party	status = no indication			
ISDN parameter	DISC: Cause value and locatio	n map	ped from	the enc	apsu	ated REL in the received			
values	CANCEL	_			-				
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	CANCEL(REL)	<b>→</b>			<b>→</b>	DISCONNECT			
	200 OK CANCEL	+			+	RELEASE			
	487 Request Terminated	+		·	<b>→</b>	RELEASE COMPLETE			
	ACK	<b>→</b>		•					

TP506007	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.11.1						
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL									
SIP selection criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message, on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side.									
SIP parameter values	183 Session Progress encapsu 183 Session Progress with enc Progress indicator									
ISDN parameter values	DISC: Cause value and locatio CANCEL	n map	ped from	the encaps	ulated REL in the received					
Comments	SIP-I		SU	T	ISDN					
	INVITE(IAM)	<b>→</b>		→	SETUP					
	183 Session Progress(ACM)	+		+	CALL PROC					
	183 Session Progress(CPG)	+		+	PROGRESS(PI)					
	CANCEL(REL)	<b>→</b>		→	DISCONNECT					
	200 OK CANCEL	+		+	RELEASE					
	487 Request Terminated	+		<b>→</b>	RELEASE COMPLETE					
	ACK	<b>→</b>		_						

TP506008	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side.								
SIP parameter values									
ISDN parameter values	DISC: Cause value and locati CANCEL	on mappe	ed from th	e encapsul	ated REL in the received				
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP				
				<b>←</b>	SETUP ACK				
	180 Ringing(ACM)	+		+	ALERTING				
	CANCEL(REL)	<b>→</b>		<b>→</b>	DISCONNECT				
	200 OK CANCEL	+	•	+	RELEASE				
	487 Request Terminated	+		<b>→</b>	RELEASE COMPLETE				
	ACK	<b>→</b>							

TP506009	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, on receipt of a ALERTING Message, on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side.								
SIP parameter values	183 Session Progress encaps	ulated AC	M: BCi C	alled party	status = no indication				
ISDN parameter	DISC: Cause value and location	n mappe	d from the	e encapsu	lated REL in the received				
values	CANCEL			·					
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP				
	183 Session Progress(ACM)	+		+	CALL PROC				
	180 Ringing(CPG)	+		+	ALERTING				
	CANCEL(REL)								
	200 OK CANCEL	+		+	RELEASE				
	487 Request Terminated	+		<b>→</b>	RELEASE COMPLETE				
	ACK	<b>→</b>	•						

TP506010	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.11.1						
TSS reference	SIP-I-ISDN/Basic_call/Receipt_	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, a ALERTING Message, a PROGRESS message, on receipt of SIP <b>CANCEL</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received CANCEL to the ISDN side.									
SIP parameter values		183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator								
ISDN parameter values	DISC: Cause value and locatio CANCEL	n map	ped from	the encar	osul	ated REL in the received				
Comments	SIP-I		SU	IT		ISDN				
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP				
	183 Session Progress(ACM)	+			<del>(</del>	CALL PROC				
	180 Ringing(CPG)	+			<del>(</del>	ALERTING				
	183 Session Progress(CPG)	+			<del>(</del>	PROGRESS(PI)				
	CANCEL(REL)	<b>→</b>			<b>→</b>	DISCONNECT				
	200 OK CANCEL	+			<del>(</del>	RELEASE				
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE				
	ACK	<b>→</b>								

TP506011	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out a SETUP message, receives an ALERTING message, the SUT on receipt of SIP <b>BYE</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side.								
SIP parameter values									
ISDN parameter values	DISC: Cause value and locati	on map	ped from	the encap	sul	ated REL in the received BYE			
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	180 Ringing(ACM)	+		,	<del>(</del>	ALERTING			
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT			
	200 OK BYE(RLC)	+			<del>(</del>	RELEASE			
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE			
	ACK	<b>→</b>							

TP506012	SIP reference: RFC 32	ISDN reference: 012.5 [1], clause 6.11.1							
TSS reference	SIP-I-ISDN/Basic_call/Receipt	of_B	YE_or_CANCE	L					
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, on receipt of SIP BYE, the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side.								
SIP parameter values	183 Session Progress encapsu	ılated	ACM: BCi Call	ed party	status = no indication				
ISDN parameter values	DISC: Cause value and location	n map	ped from the e	ncapsul	lated REL in the received BYE				
Comments	SIP-I		SUT		ISDN				
	INVITE(IAM)	<b>→</b>		<b>→</b>	SETUP				
	183 Session Progress(ACM)	+		+	CALL PROC				
	BYE(REL)								
	200 OK BYE(RLC)	200 OK BYE(RLC) ← RELEASE							
	487 Request Terminated	<b>←</b>		<b>→</b>	RELEASE COMPLETE				
	ACK	<b>→</b>							

TP506013	SIP reference: RFC 3261 [4]			ISDN reference: Q.1912.5 [1], clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL								
SIP selection									
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message followed by a PROGRESS message, on receipt of SIP <b>BYE</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side.								
SIP parameter	183 Session Progress encapsu								
values	183 Session Progress with end Progress indicator	capsulate	d CPG	Event indica	ator= Progress, ATP with				
ISDN parameter values		n mappe	d from	the encapsu	lated REL in the received BYE				
Comments	SIP-I		SU	T	ISDN				
	INVITE(IAM)	<b>→</b>		→	SETUP				
	183 Session Progress(ACM)	+		+	CALL PROC				
	183 Session Progress(CPG)	+		+	PROGRESS(PI)				
	BYE(REL)								
	200 OK BYE(RLC)	+		+	RELEASE				
	487 Request Terminated	+		<b>→</b>	RELEASE COMPLETE				
	ACK	<b>→</b>							

TP506014	SIP reference: RFC 3	3261 [4]		ISDN reference: Q.1912.5 [1], clause 6.11.1						
TSS reference	SID LISDN/Basic call/Pagain	t of BV	E or C/		Z. 1 3	12.5 [1], clause 0.11.1				
SIP selection	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL									
criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a SETUP ACKNOWLEDGE message followed by a ALERTING message, on receipt of SIP <b>BYE</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side.									
SIP parameter										
values										
ISDN parameter values	DISC: Cause value and locati	on mapp	ed from	the encap	osul	ated REL in the received BYE				
Comments	SIP-I		SU	Т		ISDN				
	INVITE(IAM)	→			<b>→</b>	SETUP				
					<del>(</del>	SETUP ACK				
	180 Ringing(ACM)	+			<del>(</del>	ALERTING				
	BYE(REL)	<b>→</b>			<b>→</b>	DISCONNECT				
	200 OK BYE(RLC)	+		Ì	<del>(</del>	RELEASE				
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE				
	ACK	<b>→</b>								

TP506015	SIP reference: RFC 32	261 [4]		ISDN reference: Q.1912.5 [1], clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receipt	_of_BYE	_or_CA	NCEL					
SIP selection criteria									
ISDN selection criteria									
Test purpose	SETUP message, receives a C Message, on receipt of SIP BY	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, on receipt of a ALERTING Message, on receipt of SIP BYE, the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side.							
SIP parameter values	183 Session Progress encaps	183 Session Progress encapsulated ACM: BCi Called party status = no indication							
ISDN parameter values	DISC: Cause value and location	n mappe	ed from	the enca	psul	ated REL in the received BYE			
Comments	SIP-I		SU	Т		ISDN			
	INVITE(IAM)	<b>→</b>			<b>→</b>	SETUP			
	183 Session Progress(ACM)	+			+	CALL PROC			
	180 Ringing(CPG)	+			<b>←</b>	ALERTING			
	BYE(REL) → DISCONNECT								
	200 OK BYE(RLC)	200 OK BYE(RLC) ← RELEASE							
	487 Request Terminated	+			<b>→</b>	RELEASE COMPLETE			
	ACK	<b>→</b>							

TP506016	SIP reference: RFC 32	261 [4	]	(	ე.19	ISDN reference: 12.5 [1], clause 6.11.1					
TSS reference	SIP-I-ISDN/Basic_call/Receipt	SIP-I-ISDN/Basic_call/Receipt_of_BYE_or_CANCEL									
SIP selection criteria											
ISDN selection criteria											
Test purpose	SETUP message, receives a C PROGRESS message, on rece	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an SETUP message, receives a CALL PROCEEDING message, a ALERTING Message, a PROGRESS message, on receipt of SIP <b>BYE</b> , the I-IWU shall send an ISDN DISC with the cause and location mapped from the encapsulated REL in the received BYE to the ISDN side									
SIP parameter values		183 Session Progress encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with									
ISDN parameter values	DISC: Cause value and locatio	n map	ped from	the enca	psul	ated REL in the received BYE					
Comments	SIP-I		SU	IT		ISDN					
	INVITE(IAM)	<b>→</b>			1	SETUP					
	183 Session Progress(ACM)	+			<b>4</b>	CALL PROC					
	180 Ringing(CPG)	+			<b>4</b>	ALERTING					
	183 Session Progress(CPG)	+			<b>+</b>	PROGRESS(PI)					
	BYE(REL)	<b>→</b>			<b>↑</b>	DISCONNECT					
	200 OK BYE(RLC)	+			+	RELEASE					
	487 Request Terminated	+			<b>↑</b>	RELEASE COMPLETE					
	ACK	<b>→</b>									

## A.1.1.1.2 Test purposes for ISDN to SIP Basic call (Incoming)

### A.1.1.2.1 Sending of the INVITE message

TP601001	SIP reference: RF	C 326	1 [4]		ISDN/ISDN reference:
				(	Q.1912.5 [1], clause 7.1 1 a)
TSS reference	ISDN-SIP /Basic call/Send	ding o	f the INVITE	message	
SIP selection					
criteria					
ISDN selection					
criteria					
Test purpose	Ensure that the SUT in Idl	e stat	e, on receipt	of a SETU	P message containing the complete
	called party number and	the <b>s</b>	ending com	<b>plete</b> indic	ation:
	<ul> <li>sends the INVITE me</li> </ul>	ssage	e.		
SIP parameter					
values					
ISDN parameter	SETUP; Called party nur	nber:	with send co	mplete ind	lication
values					
Comments	ISDN		SUT		SIP-I
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)
	ALERTING	+		+	180 Ringing(ACM)
	CONNECT	+		+	200 OK INVITE(ANM)
				<b>→</b>	ACK
			Conversat	tion	
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)
	RELEASE	+		+	200 OK BYE(RLC)
	RELEASE COMPLETE	<b>→</b>			

TP601002	SIP reference: RFC 3261 [4]				Q.	ISDN/ISDN reference: 1912.5 [1], clause 7.1 1 b)			
TSS reference	ISDN-SIP /Basic call/Send	ISDN-SIP /Basic call/Sending of the INVITE message							
SIP selection criteria									
ISDN selection criteria									
Test purpose	number of digits used in	Ensure that the SUT in Idle state, on receipt of a SETUP message containing the <b>maximum number of digits</b> used in the national numbering plan:  • sends the INVITE message.							
SIP parameter values									
ISDN parameter values	SETUP; Called party nur	nber:	complete nu	mber					
Comments	ISDN		SUT			SIP-I			
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ALERTING	+		•	<del>(</del>	180 Ringing(ACM)			
	CONNECT	+		•	<del>(</del>	200 OK INVITE(ANM)			
					<b>→</b>	ACK			
			Conversa	tion					
	DISCONNECT	<b>→</b>			<b>→</b>	BYE(REL)			
	RELEASE	+			<del>(</del>	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>							

TP601003	SIP reference: RF	C 326	1 [4]	Q	ISDN/ISDN reference: .1912.5 [1], clause 7.1 1 c)			
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message							
SIP selection criteria								
ISDN selection criteria								
Test purpose	called party number who	ere the dicate I party	e end of addr that <b>a suffici</b> :	ess signalli	JP message containing the complete ng is determined by analysis of the er of digits has been received to			
SIP parameter values								
ISDN parameter values	SETUP; Called party nu	mber:	sufficient nu	mber of dig	its to route to the called party			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONNECT	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
			Conversa	tion				
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>						

TP601004	SIP reference: RF0	C 326	61 [4]		ISDN/ISDN reference:		
					).1912.5 [1], clause 7.1 1 d)		
TSS reference	ISDN-SIP /Basic call/Send	ding o	f the INVITE n	nessage			
SIP selection criteria							
ISDN selection criteria							
Test purpose	Ensure that the SUT in Idle state, on receipt of a SETUP message containing the complete called party number where the end of address signalling is determined by the expiration timer T302 after the receipt of the latest address message:  • sends the INVITE message.						
SIP parameter							
values							
ISDN parameter							
values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		→	INVITE(IAM)		
	ALERTING	<b>←</b>		+	180 Ringing(ACM)		
	CONNECT	+		+	200 OK INVITE(ANM)		
				→	ACK		
			Conversati	on			
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)		
	RELEASE	<b>←</b>		+	200 OK BYE(RLC)		
	RELEASE COMPLETE	<b>→</b>					

TP601005	SIP reference: RF	C 3261 [4]	]	(	ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.1				
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message								
SIP selection criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT in the Idle state on receipt of an SETUP message, with the Bearer capability set to BC_VALUE:  • sends an INVITE message, with the media description defined with the "a = " "b =" and "m=" lines set to a_b_m_LINE_VALUE;  • the IAM is encapsulated unchanged in the INVITE.								
SIP parameter values									
ISDN parameter values	INVITE: a_b_m_LINE_VA	LUE, IAM	encapsul	lated in a M	IIME-body				
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
	Conversation								
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

Table 28

				Values for t	est purpose	ГР301005					
VA		IS	DN		SDP - a_b_m_LINE_VALUE						
		BC par	ameter	HLC		m= line		b= line	a= line		
	ВС	Information Transport Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification	<media></media>	<transport></transport>	<fmt-list></fmt-list>	<modifier>:<bandwi dth-value=""></bandwi></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)		
VA_02	"speech"	"Speech"	"G.711 μ-law"	Ignore	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"	"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"	"3,1 KHz audio"	"G.711 μ-law"		Audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)		
VA_06	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"		Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000		
VA_07	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 µ-law"	"Facsimile Group 2/3"	Image	udptl	t38	AS:64	Based on T.38.		
VA_08	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	Image	udptl	t38	AS:64	Based on T.38.		
VA_09	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 μ-law"	"Facsimile Group 2/3"	Image	tcptl	t38	AS:64	Based on T.38.		
VA_10	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	Image	tcptl	t38	AS:64	Based on T.38.		
VA_11	"64 kbit/s unrestricted"	"Unrestricted digital inf. W/tone/ann."	N/A	Ignore	Audio	RTP/AVP	9	AS:64	RTPmap:9 G722/8000		
VA_12	"64 kbit/s unrestricted"	"Unrestricted digital information"	N/A	Ignore	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap: <dynamic-pt> CLEARMODE/8000</dynamic-pt>		

TP601006	SIP reference: RF	C 326	1 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2				
TSS reference	ISDN-SIP /Basic call/Send	ISDN-SIP /Basic call/Sending of the INVITE message							
SIP selection									
criteria									
ISDN selection									
criteria									
Test purpose	Called Party Number para  to the addr-spec com	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the SETUP:  to the addr-spec component of the To header field which shall include the "user=phone" URI parameter if the To header field contains a sip: URI.							
SIP parameter	INVITE: To: sip:; user=								
values	1	•							
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversa	tion					
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

TP601007	SIP reference: RF	C 326	1 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2
TSS reference	ISDN-SIP /Basic call/Send	ding of	f the INVITE		2.10 1210 [1], olddod 11112
SIP selection criteria					
ISDN selection criteria					
Test purpose	Ensure that the SUT is ma Called Party Number para • to the addr-spec com	meter	of the SETL	JP and the a	
SIP parameter values	INVITE: To:	•			
ISDN parameter values					
Comments	ISDN		SUT		SIP-I
	SETUP	<b>→</b>			
	INFO	<b>→</b>			
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)
	ALERTING	+		<b>←</b>	180 Ringing(ACM)
	CONNECT	+		+	200 OK INVITE(ANM)
				→	ACK
			Conversa	tion	
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)
	RELEASE	+		+	200 OK BYE(RLC)
	RELEASE COMPLETE	<b>→</b>			

TP601008	SIP reference: RF	C 326	1 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2			
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message							
SIP selection criteria								
ISDN selection criteria								
Test purpose	Called Party address infor	matio ponen	n of the SET It of the <b>To h</b>	UP and foll eader field	which shall include the			
SIP parameter values	INVITE: To: sip:; user=							
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>						
	INFO	<b>→</b>						
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONNECT	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
			Conversa	tion				
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>						

TP601009	SIP reference: RF	C 326	1 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2			
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message							
SIP selection criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON = "International number" of the SETUP:  • to the addr-spec component of the <b>To header field</b> in the INVITE message;  • the format of the To header field is "+CC+NDC+SN";  • the forward address information is derived from the user info component of the INVITE Request-URI.							
SIP parameter values	INVITE: To:							
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	NVITE(IAM)			
	ALERTING	<b>←</b>		<del>-</del>	180 Ringing(ACM)			
	CONNECT	+		+	200 OK INVITE(ANM)			
				→	ACK ACK			
			Conversat	ion				
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>						

TP601010	SIP reference: RF0	C 326	1 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2					
TSS reference	ISDN-SIP /Basic call/Send								
SIP selection	ISDN-SIP /Basic call/Sending of the INVITE message								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON = "National (significant) number" of the SETUP:  to the addr-spec component of the To header field in the INVITE message;  the format of the To header field is "+CC+NDC+SN";  the forward address information is derived from the user info component of the INVITE Request-URI.								
SIP parameter values	INVITE: To:								
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		7	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	<b>←</b>		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversat	tion					
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		€	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

TP601011	SIP reference: RFC 3261 [4]				(	ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2	
TSS reference	ISDN-SIP /Basic call/Send	ling o	f the INVITE	messag	е		
SIP selection criteria							
ISDN selection criteria							
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON = "unknown" of the SETUP:  to the addr-spec component of the To header field in the INVITE message;  the format of the To header field is "+CC+NDC+SN";  the forward address information is derived from the user info component of the INVITE Request-URI.						
SIP parameter values	INVITE: To:						
ISDN parameter values							
Comments	ISDN		SUT			SIP-I	
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)	
	ALERTING	+			<b>←</b>	180 Ringing(ACM)	
	CONNECT	+			<del>(</del>	200 OK INVITE(ANM)	
					<b>→</b>	ACK	
			Conversat	ion			
	DISCONNECT	<b>→</b>			<b>→</b>	BYE(REL)	
	RELEASE	+			<del>(</del>	200 OK BYE(RLC)	
	RELEASE COMPLETE	<b>→</b>	_				

TP601012	SIP reference: RF	eference: RFC 3261 [4]			ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2				
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON = "International number" of the SETUP and the and the followed INFO:  • to the addr-spec component of the To header field;  • the format of the To header field is "+CC+NDC+SN";  • the forward address information is derived from the user info component of the INVITE Request-URI.								
SIP parameter values	INVITE: To:								
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>							
	INFO	<b>→</b>							
	INFO	<b>→</b>		→	INVITE(IAM)				
	ALERTING	<b>←</b>		<b>←</b>	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversa	tion					
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

TP601013	SIP reference: RF	C 326	1 [4]		ISDN/ISDN reference:					
TSS reference	Q.1912.5 [1], clause 7.1.2									
SIP selection	ISDN-SIP /Basic call/Send	ISDN-SIP /Basic call/Sending of the INVITE message								
criteria										
ISDN selection										
criteria										
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON = "National (significant) number" of the SETUP and the followed INFO:  • to the addr-spec component of the To header field;  • the format of the To header field is "+CC+NDC+SN";  • the forward address information is derived from the user info component of the INVITE Request-URI.									
SIP parameter	INVITE: To:									
values										
ISDN parameter values										
Comments	ISDN		SUT		SIP-I					
	SETUP	<b>→</b>								
	INFO	<b>→</b>								
	INFO	<b>→</b>		→	INVITE(IAM)					
	ALERTING	+		<b>←</b>	180 Ringing(ACM)					
	CONNECT	+		+	200 OK INVITE(ANM)					
				→	ACK					
			Conversa	tion						
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)					
	RELEASE	+		+	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>→</b>								

TP601014	SIP reference: RFC 3261 [4]				(	ISDN/ISDN reference: Q.1912.5 [1], clause 7.1.2			
TSS reference	ISDN-SIP /Basic call/Sending of the INVITE message								
SIP selection									
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, TON = "unknown" of the SETUP and the followed INFO:  to the addr-spec component of the To header field;  the format of the To header field is "+CC+NDC+SN";  the forward address information is derived from the user info component of the INVITE Request-URI.								
SIP parameter	INVITE: To:								
values									
ISDN parameter values									
Comments	ISDN		SUT			SIP-I			
	SETUP	<b>→</b>							
	INFO	<b>→</b>							
	INFO	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ALERTING	<b>←</b>			<del>(</del>	180 Ringing(ACM)			
	CONNECT	<b>←</b>			<b>←</b>	200 OK INVITE(ANM)			
					<b>→</b>	ACK			
			Conversa	tion					
	DISCONNECT	<b>→</b>			<b>→</b>	BYE(REL)			
	RELEASE	+			<b>←</b>	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>							

#### A.1.1.2.2 Overlap sending

TP602001	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.2				
TSS reference	ISDN-SIP /Basic call/Over	lap se	ending						
SIP selection criteria	PICS 3/1	PICS 3/1							
ISDN selection criteria									
Test purpose		Ensure if the SUT is supporting en bloc addressing towards the SIP network, subsequent INFOs received after the SUT has sent the INVITE are ignored.							
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	INFO	<b>→</b>			·				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversat	ion					
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

TP602002	SIP reference: RF	C 3261 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.2.1							
TSS reference	ISDN-SIP /Basic call/Ove	ISDN-SIP /Basic call/Overlap sending									
SIP selection	PICS 3/2										
criteria											
ISDN selection											
criteria											
Test purpose	message. On receipt of a  1) Stop timer TOIW3 (if it  2) TOIW2 shall be restar  a) The Request-URI  received so far for  b) A new INVITE with previous INVITE is  c) The new INVITE s  resources that hav reserved resources parameters in ques	Ensure that the SUT in Idle state, on receipt of an SETUP message sends an INVITE message. On receipt of a INFO from the ISDN access the SUT shall:  1) Stop timer TOIW3 (if it is running); 2) TOIW2 shall be restarted and the SUT shall invoke the following procedures: a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call; b) A new INVITE with the INFOe Call-ID and From header (including tag) as the previous INVITE is sent; c) The new INVITE shall contain a new SDP offer. The O-IWU 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 SETUP.									
SIP parameter values											
ISDN parameter											
values		T		I							
Comments	ISDN	SI		SIP-I							
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)							
	INFO	<b>→</b>	<b>→</b>	INVITE(IAM)							
	INFO	<b>→</b>	→	INVITE(IAM)							
		<u> </u>									
	ALERTING	<del>-</del>	<del>-</del>	180 Ringing(ACM)							
	CONNECT	+	+	200 OK INVITE(ANM)							
			→	ACK							
		Conve									
	DISCONNECT	<b>→</b>	→	BYE(REL)							
1	RELEASE ← 200 OK BYE(RLC)										
		RELEASE COMPLETE +									

TP602003	SIP reference: RFC 3261 [4]			(	ISDN/ISDN reference: Q.1912.5 [1], clause 7.2.1		
TSS reference	ISDN-SIP /Basic call/Over	lap se	ending				
SIP selection criteria	PICS 3/2						
ISDN selection criteria							
Test purpose	The SUT in Idle state, on receipt of an SETUP message sends a INVITE message On receipt of a INFO from the ISDN access the SUT shall:  TOIW2 shall be restarted and the SUT shall invoke the following procedures: Ensure that if timer TOIW2 has expired, subsequent INFOs received after the SUT has sent the INVITE are ignored.						
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	INFO	<b>→</b>		→	INVITE(IAM)		
			T <sub>oiw2</sub> expi	ired			
	INFO	<b>→</b>					
	ALERTING	+		<b>+</b>	180 Ringing(ACM)		
	CONNECT	+		<del>-</del>	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
			Conversa	tion			
	DISCONNECT	<b>→</b>		→	BYE(REL)		
	RELEASE	+		+	200 OK BYE(RLC)		
	RELEASE COMPLETE	<b>→</b>					

TP602004	SIP reference: RF	C 326	61 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.2.1				
TSS reference	ISDN-SIP /Basic call/Ove	ISDN-SIP /Basic call/Overlap sending							
SIP selection criteria	PICS 3/1								
ISDN selection criteria									
Test purpose	<ul> <li>The SUT in Idle state, on receipt of a SETUP message. On receipt of a INFO from the BICC/ISDN the SUT shall:</li> <li>sends an INVITE message if the minimum number of digits for routing the call has been received in the SETUP and the INFO TOIW1 and TIOW2 shall be started and the SUT shall invoke the following procedures;</li> <li>ensure that if timer TOIW2 has expired, subsequent INFOs received after the SUT has sent the INVITE are ignored.</li> </ul>								
SIP parameter values									
ISDN parameter									
values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>							
	INFO	<b>→</b>		→	INVITE(IAM)				
			T <sub>oiw2</sub> expi	ired					
	INFO	<b>→</b>							
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversa	tion					
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

TP602005	SIP reference: RFC 3261 [4]				ISDN/ISDN reference: Q.1912.5 [1], clause 7.2.1					
TSS reference	ISDN-SIP /Basic call/Ove	rlap se	ending							
SIP selection criteria	PICS 1/9 AND PICS 3/2									
ISDN selection criteria										
Test purpose	Ensure that if the O-MGCF sends an INVITE request before the end of address signalling is determined, the O-MGCF shall an INVITE with incomplete address information reject with a SIP 404 or 484 error response.  On receipt of a INFO from the ISDN access, the O-MGCF shall: stop timer Ti/w3 (if it is running); send an INVITE request complying to the following:  - the INVITE request shall use the SIP preconditions extension;  - the INVITE request shall include all digits received so far for this call in the Request-URI;  - restart Ti/w2.									
SIP parameter values										
ISDN parameter values										
Comments	ISDN		SUT		SIP-I					
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)					
				+	404/484					
				<b>→</b>	ACK					
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)					
				+	404/484					
				<b>→</b>	ACK					
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)					
				+	404/484					
				<b>→</b>	ACK					
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ALERTING	+		<b>←</b>	180 Ringing(ACM)					
	CONNECT	+		+	200 OK INVITE(ANM)					
	-			<b>→</b>	ACK					
			Conversa							
	DISCONNECT	<b>→</b>		→	BYE(REL)					
	RELEASE	+		+	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>→</b>								

TP602006	SIP reference: RFC 32	61 [4]		ISDN/ISDN reference: .1912.5 [1], clause 7.7.6					
TSS reference	ISDN-SIP /Basic call/Overlap sending								
SIP selection criteria	NOT PICS 3/2								
ISDN selection criteria									
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA:  • sends a DISCONNECT or RELEASE message cause value 28.								
SIP parameter values	SIP_Failure_VA: ISUP REL e	ncapsulated	I in the MIME I	oody					
ISDN parameter values	DISCONNECT/RELEASE: Ca	ause value o	onstructed fro	m the encapsulated REL					
Comments	ISDN		SUT	SIP-I					
	SETUP	<b>→</b>	-3	INVITE(IAM)					
			•	484 Address Incomplete					
	CASE A		-	ACK					
	RELEASE	<del>-</del>							
	RELEASE COMPLETE	<b>→</b>							
	CASE B								
	DISCONNECT	<del>-</del>							
	RELEASE	<b>→</b>							
	RELEASE COMPLETE	+							

# A.1.1.2.3 Sending of the CALL PROCEEDING / ALERTING message

TP603001	SIP reference: RF	C 326	1 [4]	Q 19 <sup>2</sup>	ISDN/ISDN reference: 12.5 [1], clauses 7.1 1) a), 7.3.1				
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING								
SIP selection criteria	PICS 3/1								
ISDN selection criteria									
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete called party number and the sending complete indication:  sends the INVITE message to called user; sends the CALL PROCEDING message; sends the PROGRESS message, with the with progress description set to PI_VAL.								
SIP parameter values	183 Session Progress with encapsulated ACM: BCi Called party status = no indication 183 Session Progress with encapsulated CPG Event indicator= Progress, ATP with Progress indicator								
ISDN parameter values	CALL PROCEEDING PROGRESS(PI value=PI	VAL)							
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	CALL PROCEEDING	+		+	183 Session Progress(ACM)				
	PROGRESS(PI)	+		+	183 Session Progress(CPG)				
	ALERTING	+		+	180 Ringing(ACM				
	CONNECT	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversa						
	DISCONNECT	→		→	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

TP603002	SIP reference: RF	C 326	1 [4]	Q.191	ISDN/ISDN reference: 2.5 [1], clauses 7.1 1) b), 7.3.1				
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING								
SIP selection	PICS 3/1								
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the maximum number of digits used in the national numbering plan:  sends the INVITE message to called user; sends the CALL PROCEDING message, with the with progress description value set to PI VAL.								
SIP parameter	183 Session Progress with	n enca	apsulated AC	M: BCi Call	ed party status = no indication, ATP				
values	with PI								
ISDN parameter	CALL PROCEEDING(PI v	alue=	PI_VAL)						
values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		→	INVITE(IAM)				
	CALL PROCEEDING	+		<b>←</b>	183 Session Progress(ACM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONNECT	<b>←</b>		←	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversat	ion					
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

TP603003	SIP reference: RF	C 326	61 [4]	Q.1	ISDN/ISDN reference: 1912.5 [1], clauses 7.1, 7.3.1			
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING							
SIP selection criteria	PICS 3/2							
ISDN selection criteria								
Test purpose	Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an SETUP message containing the minimum number of digits required for routing the call has been received (start timer TOIW2 and invoke the appropriate outgoing SIP signalling procedure)  • Sends an INVITE message to the called user  • after the expiration of T <sub>OIW2</sub> sends the CALL PROCEEDING message							
SIP parameter values								
ISDN parameter values	CALL PROCEEDING							
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>						
	INFO	<b>→</b>		→	INVITE(IAM)			
			T <sub>oiw2</sub> expired	t				
	CALL PROCEEDING	+						
	ALERTING	+		+	180 Ringing(ACM)			
	CONNECT	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
			Conversa	tion				
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>						

TP603004	SIP reference: RF	C 326	1 [4]	0 101	ISDN/ISDN reference: 12.5 [1], clauses 7.1 1) a), 7.3.1			
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING							
SIP selection criteria	PICS 3/1	<u>g_</u> c	<u> </u>	<u> </u>	LEKTING			
ISDN selection criteria								
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete called party number and the sending complete indication, on receipt of a 180 Ringing message:  step states and the sending complete indication, on receipt of a 180 Ringing message:  states and the ALERTING message with the with the with progress description value PI VAL.							
SIP parameter values	180 Ringing encapsulated Progress indicator PI_VAL		: BCi Called	party status	s = subscriber free, ATP with			
ISDN parameter values	ALERTING: Progress indi	cator	value PI_VAI	_ included				
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CALL PROCEEDING	+						
	ALERTING	+		<b>+</b>	180 Ringing(ACM(PI))			
	CONNECT	<b>←</b>		<b>+</b>	200 OK INVITE(ANM)			
				→	ACK			
			Conversat	tion				
	DISCONNECT	<b>→</b>		→	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>	_					

TP603005	SIP reference: RF	C 326	1 [4]	Q.	ISDN/ISDN reference: .1912.5 [1], clause 7.1 1 a)				
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING								
SIP selection	PICS 3/2								
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete called party number where the end of address signalling is determined by the expiration timer T <sub>OIW2</sub> after the receipt of the latest address message on receipt of a 183 Session								
	Progress with encapsulat <ul><li>a PROGRESS is ser</li></ul>	ed AC	:M: kward.						
SIP parameter		capsu	lated ACM: BC	Ci Called pa	arty status = no indication, ATP with				
values	Progress indicator								
ISDN parameter	PROGRESS								
values					T				
Comments	ISDN		SUT		SIP-I				
	SETUP	→							
	INFO	→		→	INVITE(IAM)				
			T <sub>oiw2</sub> expire	ed					
	CALL PROCEEDING	+							
	PROGRESS	+		+	183 Session Progress(ACM(PI))				
	ALERTING	+		+	180 Ringing(CPG)				
	CONNECT	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Conversati	on					
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

TP603006	SIP reference: RF0	C 326	1 [4]	C	ISDN/ISDN reference: 0.1912.5 [1], clause 7.1 1 a)				
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING								
SIP selection criteria	PICS 3/2								
ISDN selection criteria									
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete called party number where the end of address signalling is determined by the expiration timer T <sub>OIW2</sub> after the receipt of the latest address message on receipt of a 183 Session Progress:  • no information is sent backward.								
SIP parameter values	183 Session Progress end ATP	apsul	ated ACM: B	Ci Called p	party status = no indication, without				
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>							
	INFO	<b>→</b>		<b>→</b>	INVITE(IAM)				
			T <sub>oiw2</sub> expi	red					
				+	183 Session Progress(ACM)				
	ALERTING	+		+	180 Ringing(ACM				
	CONNECT	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conversat	ion					
	DISCONNECT	<b>→</b>		→	BYE(REL)				
	RELEASE	<b>←</b>		<b>←</b>	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							

TP603007	SIP reference: RF0	326	1 [4]		Q	ISDN/ISDN reference: .1912.5 [1], clause 7.1 1 a)			
TSS reference	ISDN-SIP /Basic call/Sending_of_CALL PROCEDING_ALERTING								
SIP selection criteria	PICS 3/2								
ISDN selection criteria									
Test purpose	Ensure that the SUT in Idle state, on receipt of an SETUP message containing the complete <b>called party number</b> where the end of address signalling is determined by the expiration timer T <sub>OIW2</sub> after the receipt of the latest address message on receipt of a 183 Session Progress:  • no information is sent backward.								
SIP parameter values	183 Session Progress with	nout e	ncapsulated	ISUP me	ess	age			
ISDN parameter values									
Comments	ISDN		SUT			SIP-I			
	SETUP	<b>→</b>							
	INFO	<b>→</b>		-	<del>&gt;</del>	INVITE(IAM)			
			T <sub>oiw2</sub> expi	red					
				•	<del>(</del>	183 Session Progress			
	ALERTING	+		•	<del>(</del>	180 Ringing(ACM			
	CONNECT	+		•	<del>(</del>	200 OK INVITE(ANM)			
				-	<b>→</b>	ACK			
		·	Conversat	ion					
	DISCONNECT	<b>→</b>			<b>→</b>	BYE(REL)			
	RELEASE	+		•	<del>(</del>	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>							

## A.1.1.2.4 Sending of the CONNECT message

TP604001	SIP reference: RF	C 326		ISDN/ISDN reference: Q.1912.5 [1], clause 7.5							
TSS reference	ISDN-SIP /Basic call/Send	ding_o	of_CONNECT								
SIP selection criteria											
ISDN selection criteria											
Test purpose	for this call, it shall stop tir <ul><li>send CONNECT as d</li></ul>	Ensure that the SUT having sent the ALERTING message, on receipt of a 200 OK INVITE for this call, it shall stop timer T <sub>OIW2</sub> (if running):  • send CONNECT as determined by ISDN procedures;  • stop any existing awaiting answer indication (e.g. ringing tone).									
SIP parameter values	200 OK INVITE: encapsul	ated A	ANM in the MII	ME body							
ISDN parameter values	CONNECT										
Comments	ISDN		SUT		SIP-I						
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)						
	ALERTING	+		+	180 Ringing(ACM						
	CONNECT	+		+	200 OK INVITE(ANM)						
				<b>→</b>	ACK						
			Conversati	on							
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)						
	RELEASE	+		+	200 OK BYE(RLC)						
	RELEASE COMPLETE	<b>→</b>									

TP604002	SIP reference: RF	C 326	1 [4]			ISDN/ISDN reference: Q.1912.5 [1], clause 7.5			
TSS reference	ISDN-SIP /Basic call/Sen	ding c	of_CONNECT	Ī					
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT does not having sent the ALERTING message, on receipt of a 200 OK INVITE for this call, it shall stop timer T <sub>OIW2</sub> (if running):  send CON as determined by ISDN procedures; Sstop any existing awaiting answer indication (e.g. ringing tone).								
SIP parameter values	200 OK INVITE: encapsu					3 3 ,			
ISDN parameter values	CONNECT								
Comments	ISDN		SUT			SIP-I			
	SETUP	<b>→</b>		-	<del>}</del>	INVITE(IAM)			
	CONNECT	+		•	<del>(</del>	200 OK INVITE(CON)			
				-	<del>&gt;</del>	ACK			
			Conversat	tion					
	DISCONNECT	<b>→</b>		-	<del>&gt;</del>	BYE(REL)			
	RELEASE	+		•	<del>-</del>	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>							

### A.1.1.2.5 Receipt of the RELEASE or DISCONNECT

TP605001	SIP reference: RF	C 326	1 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1, 1)			
TSS reference	ISDN-SIP /Basic call/Rece	eipt_o	f_RELEASE	or_DISC	ONNECT			
SIP selection								
criteria								
ISDN selection criteria								
Test purpose	receipt of a RELEASE CC	MPLI	ETĔ messag	e:	efore an INVITE has been sent. On terminate local procedures if any are			
SIP parameter values								
ISDN parameter values	RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)							
Comments	ISDN	ISDN SUT SIP-I						
	SETUP	<b>→</b>						
	RELEASE COMPLETE	<b>→</b>						

TP605002	SIP reference: RF0	3261 [4]	Q.	ISDN/ISDN reference: 1912.5 [1], clause 7.7.1, 1)						
TSS reference	ISDN-SIP /Basic call/Rece	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection criteria										
ISDN selection criteria										
Test purpose	Ensure that the SUT after receiving the SETUP but before an INVITE has been sent. On receipt of a RELEASE message:  on action is required on the SIP side other than to terminate local procedures if any are in progress;									
SIP parameter values										
ISDN parameter values	RELEASE; cause value: (	(PIXIT), location	(PIXIT)							
Comments	ISDN	SUT		SIP-I						
	SETUP	<b>→</b>								
	RELEASE	<b>→</b>								
	RELEASE COMPLETE	+								

TP605003	SIP reference: RF0	C 326	1 [4]	(	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1, 1)				
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT after receiving the SETUP but before an INVITE has been sent. On receipt of a DISCONNECT message:  no action is required on the SIP side other than to terminate local procedures if any are in progress.								
SIP parameter values									
ISDN parameter values	DISCONNECT; cause val	lue: (l	PIXIT), locat	ion (PIXIT					
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>							
	DISCONNECT	<b>→</b>							
	RELEASE	+							
	RELEASE COMPLETE →								

TP605004	SIP reference: RF	C 326	1 [4]	Q	ISDN/ISDN reference: .1912.5 [1], clause 7.7.1 2)				
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection criteria	NOT PICS 4/10								
ISDN selection criteria									
Test purpose  SIP parameter	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE ISDN message before a 200 OK (any) response message has been received which establishes a confirmed dialogue:  • the SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received;  • the SUT shall send a BYE request.								
values									
ISDN parameter values	RELEASE COMPLETE; c	ause	value: (PIXI	T), location	(PIXIT)				
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	RELEASE COMPLETE	<b>→</b>							
				+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
				→	BYE(REL)				
				←	200 OK BYE(RLC)				

TP605005	SIP reference: RF	C 3261 [4]		ISDN/ISDN reference:			
T00 (	Q.1912.5 [1], clause 7.7.1 2)						
TSS reference	ISDN-SIP /Basic call/Rece	eipt_of_RELEASI	_or_DISCO	NNECT			
SIP selection	NOT PICS 4/10						
criteria							
ISDN selection							
criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE ISDN message <b>before</b> a 200 OK (any) response message has been received which establishes a confirmed dialogue:  • the SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received;  • the SUT shall send a BYE request.						
SIP parameter values							
ISDN parameter values	RELEASE; cause value: (PIXIT), location (PIXIT)						
Comments	ISDN	SUT		SIP-I			
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)			
	RELEASE	<b>→</b>					
	RELEASE COMPLETE (+						
	← 200 OK INVITE(ANM)						
	→ ACK						
	→ BYE(REL)						
	← 200 OK BYE(RLC)						

TP605006	SIP reference: RF	C 3261 [4]	Q	ISDN/ISDN reference: .1912.5 [1], clause 7.7.1 2)		
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT					
SIP selection	NOT PICS 4/10					
criteria						
ISDN selection						
criteria						
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message with Cause value CV_ISDN, location LOC_ISDN before a 200 OK (any) response message has been received which establishes a confirmed dialogue:  • the SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received;  • the SUT shall send a BYE request.					
SIP parameter values						
ISDN parameter values	DISCONNECT; cause value: (PIXIT), location (PIXIT)					
Comments	ISDN SUT SIP-I					
	SETUP → INVITE(IAM)					
	DISCONNECT →					
	RELEASE ← 200 OK INVITE(ANM)					
			→	ACK		
			→	BYE(REL)		
			<b>←</b>	200 OK BYE(RLC)		

TP605007	SIP reference: RF	C 3261	[4]	Q.1	ISDN/ISDN reference: 1912.5 [1], clause 7.7.1 2) 3)
TSS reference	ISDN-SIP /Basic call/Rece	eipt_of_	RELEASE	or_DISCOI	NNECT
SIP selection	NOT PICS 4/10				
criteria					
ISDN selection					
criteria					
Test purpose  SIP parameter	<ul> <li>Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE ISDN message before a 200 OK SIP response message has been received:</li> <li>the SUT shall hold the release procedure. A CANCEL is sent when any SIP response was 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> </ul>				
values					
ISDN parameter values	RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)				
Comments	ISDN		SUT		SIP-I
	SETUP	<b>→</b>		→	INVITE(IAM)
				+	100 Trying
	RELEASE COMPLETE	→			
				→	CANCEL(REL)
				<b>←</b>	200 OK INVITE(ANM)
				→	ACK
				+	200 OK CANCEL
				→	BYE(REL)
				<b>←</b>	200 OK BYE(RLC)

TP605008	SIP reference: RF	C 3261 [4]		Q.1	ISDN/ISDN reference: 912.5 [1], clause 7.7.1 2) 3)
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT				
SIP selection	NOT PICS 4/10				
criteria					
ISDN selection					
criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE message ISDN <b>before</b> a 200 OK SIP response message has been received:				
	<ul> <li>the SUT shall hold the was been received;</li> </ul>	e release p	rocedur	e. A CANCI	EL is sent when any SIP response
					es, the SUT shall send an ACK for request after the ACK has been sent.
SIP parameter		•	•		•
values					
ISDN parameter values	RELEASE; cause value: (PIXIT), location (PIXIT)				
Comments	ISDN		SUT		SIP-I
	SETUP	<b>→</b>		→	INVITE(IAM)
				+	100 Trying
	RELEASE	<b>→</b>			
	RELEASE COMPLETE	+		<b>→</b>	CANCEL(REL)
				+	200 OK INVITE(ANM)
				<b>→</b>	ACK
				<b>←</b>	200 OK CANCEL
				<b>→</b>	BYE(REL)
		<b>→</b>		+	200 OK BYE(RLC)

TP605009	SIP reference: RF	C 3261 [4]		Q.1	ISDN/ISDN reference: 912.5 [1], clause 7.7.1 2) 3)	
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT					
SIP selection	NOT PICS 4/10					
criteria						
ISDN selection criteria						
Test purpose	<ul> <li>Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message ISDN before a 200 OK SIP response message has been received:</li> <li>the SUT shall hold the release procedure. A CANCEL is sent when any SIP response was 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> </ul>					
SIP parameter values						
ISDN parameter values	DISCONNECT; cause value: (PIXIT), location (PIXIT)					
Comments	ISDN	S	UT		SIP-I	
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)	
				<del>(</del>	100 Trying	
	DISCONNECT → CANCEL(REL)  RELEASE ← → CANCEL(REL)  RELEASE COMPLETE → ← 200 OK INVITE(ANM)					
				<b>→</b>	ACK	
				<del>(</del>	200 OK CANCEL	
				<b>→</b>	BYE(REL)	
				<b>←</b>	200 OK BYE(RLC)	

TP605010	SIP reference: RF	C 326	1 [4]	Q.1	ISDN/ISDN reference: 912.5 [1], clause 7.7.1 2) 3)		
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT						
SIP selection criteria	NOT PICS 4/10						
ISDN selection criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message ISDN message <b>before</b> an early dialogue with the message defined as SIP_MESSAGE_VA has been established:  • the SUT shall hold the release procedure until a <b>SIP_MESSAGE_VA</b> response has been received;  • the SUT shall send a CANCEL or BYE request.						
SIP parameter values							
ISDN parameter values	RELEASE COMPLETE; cause value: (PIXIT), location (PIXIT)						
Comments	ISDN SUT SIP-I						
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	100 Trying		
	RELEASE COMPLETE	<b>→</b>					
				+	SIP_MESSAGE_VA		
				CAS	E A		
	→ CANCEL(REL)						
				+	200 OK CANCEL		
	← 487 Request Terminated						
				<b>→</b>	ACK		
	CASE B						
					BYE(REL)		
	← 200 OK BYE(RLC)						
				+	487 Request Terminated		
				→	ACK		

TP605011	SIP reference: RF	C 3261 [4]	Q.1	ISDN/ISDN reference: 1912.5 [1], clause 7.7.1 2) 3)			
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT						
SIP selection criteria	NOT PICS 4/10						
ISDN selection criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE ISDN message before an early dialogue with the message defined as SIP_MESSAGE_VA has been established:  • the SUT shall hold the release procedure until a SIP_MESSAGE_VA response has been received;  • the SUT shall send a CANCEL or BYE request.						
SIP parameter values							
ISDN parameter values	RELEASE; cause value:	(PIXIT), location	(PIXIT)				
Comments	ISDN SUT SIP-I						
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)			
			<b>←</b>	100 Trying			
	RELEASE	<b>→</b>					
	RELEASE COMPLETE	<del>-</del>					
			<b>←</b>	SIP_MESSAGE_VA			
				CASE A			
	→ CANCEL(REL)						
	CASE B						
			<b>→</b>	BYE(REL)			
			<b>+</b>	200 OK BYE(RLC)			
	← 487 Request Terminated						
		<u> </u>	<b>→</b>	ACK			

TP605012	SIP reference: RF	C 326	1 [4]	Q.1	ISDN/ISDN reference: 1912.5 [1], clause 7.7.1 2) 3)		
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT						
SIP selection criteria	NOT PICS 4/10						
ISDN selection criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message ISDN before an early dialogue with the message defined as SIP_MESSAGE_VA has been established:  • the SUT shall hold the release procedure until a SIP_MESSAGE_VA response has been received;  • the SUT shall send a CANCEL or BYE request.						
SIP parameter values							
ISDN parameter values	DISCONNECT; cause va	lue: (F	PIXIT), locati	ion (PIXIT)			
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	100 Trying		
	DISCONNECT	<b>→</b>					
	RELEASE	+					
	RELEASE COMPLETE	<b>→</b>					
				<b>+</b>	SIP_MESSAGE_VA		
	CASE A						
				<b>→</b>	CANCEL(REL)		
				<b>+</b>	200 OK CANCEL		
				<del>-</del>	487 Request Terminated		
				→	ACK		
	CASE B						
				<b>→</b>	BYE(REL)		
				+	200 OK BYE(RLC)		
				+	487 Request Terminated		
				<b>→</b>	ACK		

TP605013	SIP reference: RFC 3261 [4]			ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 4)			
TSS reference	ISDN-SIP /Basic call/Rece	eipt_o	f_RELEASE	_or_DISCO	NNECT		
SIP selection criteria	NOT PICS 4/10						
ISDN selection criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message ISDN message after a 200 OK response message has been received:  • the SUT shall hold the release procedure until an ACK has been sent;  • the SUT shall send a BYE request.						
SIP parameter values							
ISDN parameter values	RELEASE COMPLETE; c	ause	value: (PIXI	T), location	n (PIXIT)		
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ALERTING	+		<b>←</b>	180 Ringing(ACM		
	CONNECT	+		+	200 OK INVITE(ANM)		
	RELEASE COMPLETE	<b>→</b>					
				<b>→</b>	ACK		
				<b>→</b>	BYE(REL)		
				+	200 OK BYE(RLC)		

TP605014	SIP reference: RF	C 3261 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 4)				
TSS reference	ISDN-SIP /Basic call/Rec	eipt_of_RELEASE	_or_DISCO	NNECT			
SIP selection	NOT PICS 4/10						
criteria							
ISDN selection							
criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE ISDN message after a 200 OK response message has been received  The SUT shall hold the release procedure until an ACK has been sent  The SUT shall send a BYE request						
SIP parameter values							
ISDN parameter values	RELEASE; cause value:	(PIXIT), location (	(PIXIT)				
Comments	ISDN	SUT		SIP-I			
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)			
	ALERTING	<b>←</b>	<b>←</b>	180 Ringing(ACM			
	CONNECT	<b>←</b>	+	200 OK INVITE(ANM)			
	RELEASE	→					
	RELEASE COMPLETE	<b>←</b>	<b>→</b>	ACK			
			<b>→</b>	BYE(REL)			
			+	200 OK BYE(RLC)			

TP605015	SIP reference: RF	C 3261 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 4)				
TSS reference	ISDN-SIP /Basic call/Rece	eipt_of_RELEASE	_or_DISCOI	NNECT			
SIP selection	NOT PICS 4/10						
criteria							
ISDN selection							
criteria							
SIP parameter	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT ISDN message after a 200 OK response message has been received:  • the SUT shall hold the release procedure until an ACK has been sent;  • the SUT shall send a BYE request.						
ISDN parameter values	DISCONNECT; cause val	lue: (PIXIT), loca	tion (PIXIT)				
Comments	ISDN	SUT		SIP-I			
	SETUP	<b>→</b>	→	INVITE(IAM)			
	ALERTING	<del>-</del>	+	180 Ringing(ACM			
	CONNECT	<del>-</del>	<b>←</b>	200 OK INVITE(ANM)			
	DISCONNECT	<b>→</b>					
	RELEASE	<del>-</del>	<b>→</b>	ACK			
	RELEASE COMPLETE	<b>→</b>	<b>→</b>	BYE(REL)			
			+	200 OK BYE(RLC)			

TP605016	SIP reference: RF			ISDN/ISDN reference: .1912.5 [1], clause 7.7.1 3)
TSS reference	ISDN-SIP /Basic call/Rec	eipt_of_RELEASE	_or_DISCO	NNECT
SIP selection	NOT PICS 4/10			
criteria				
ISDN selection				
criteria				
Test purpose				complete called party number,
				COMPLETE message ISDN
	message after an early d		P message	defined with the
	SIP_MESSAGE_VA has			
	<ul> <li>the SUT shall send a</li> </ul>	CANCEL or BYE	request.	
SIP parameter				
values				
ISDN parameter	RELEASE COMPLETE;	ause value: (PIXI	T), location	(PIXIT)
values			ı	I
Comments	ISDN	SUT		SIP-I
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)
			<b>←</b>	SIP_MESSAGE_VA
	RELEASE COMPLETE	→		
	CASE A			
			<b>→</b>	CANCEL(REL)
			<b>←</b>	200 OK CANCEL
			+	487 Request Terminated
			<b>→</b>	ACK
	CASE B			
			→	BYE(REL)
			+	200 OK BYE(RLC)
			+	487 Request Terminated
			<b>→</b>	ACK

TP605017	SIP reference: RFC 3261 [4]			ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)			
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT						
SIP selection	NOT PICS 4/10						
criteria							
ISDN selection							
criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE message ISDN message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established:  • the SUT shall send a CANCEL or BYE request;						
SIP parameter values							
ISDN parameter values	RELEASE; cause value:	(PIXIT), I	location (	PIXIT)			
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		→	INVITE(IAM)		
				<b>←</b>	SIP_MESSAGE_VA		
	RELEASE	→					
	RELEASE COMPLETE	+					
	CASE A						
				→	CANCEL(REL)		
				+	200 OK CANCEL		
				+	487 Request Terminated		
				→	ACK		
	CASE B						
				→	BYE(REL)		
				+	200 OK BYE(RLC)		
				<b>←</b>	487 Request Terminated		
				→	ACK		

TP605018	SIP reference: RFC 3261 [4]			ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)				
TSS reference	ISDN-SIP /Basic call/Rec	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT						
SIP selection	NOT PICS 4/10							
criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message ISDN message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established:  • the SUT shall send a CANCEL or BYE request.							
SIP parameter values								
ISDN parameter values	DISCONNECT; cause va	lue: (F	PIXIT), <b>locat</b> i	on (PIXIT)				
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		→	INVITE(IAM)			
				+	SIP_MESSAGE_VA			
	DISCONNECT	<b>→</b>						
	RELEASE	+						
	RELEASE COMPLETE	<b>→</b>						
	CASE A							
				→	CANCEL(REL)			
				+	200 OK CANCEL			
				+	487 Request Terminated			
				→	ACK			
	CASE B							
				<b>→</b>	BYE(REL)			
				<del>(</del>	200 OK BYE(RLC)			
			<u> </u>	+	487 Request Terminated			
				<b>→</b>	ACK			

TP605019	SIP reference: RF0	326	1 [4]		Q.1	ISDN/ISDN reference: 912.5 [1], clause 7.7.1 2) 4)			
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection criteria	PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message on receipt RELEASE COMPLETE message with Cause value CV_ISDN, location LOC_ISDN before a 200 OK response (any) message has been received which establishes a confirmed dialogue:  • the SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received;  • the SUT shall send a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.								
SIP parameter values	BYE:Reason header value RELEASE COMPLETE	CV_	SIP, encapsı	ulated R	EL (	constructed from the ISDN			
ISDN parameter values	RELEASE COMPLETE; c	ause	value: (PIXI	T), loca	tion	(PIXIT)			
Comments	ISDN		SUT			SIP-I			
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)			
	RELEASE COMPLETE	<b>→</b>							
						200 OK INVITE(ANM)			
					<b>→</b>	ACK			
						BYE(REL)			
					<del>(</del>	200 OK BYE(RLC)			

TP605020	SIP reference: RF0	326	1 [4]		Q.1	ISDN/ISDN reference: 912.5 [1], clause 7.7.1 2) 4)			
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection	PICS 4/10								
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message on receipt RELEASE message with Cause value CV_ISDN, location LOC_ISDN before a 200 OK response (any) message has been received which establishes a confirmed dialogue:  • the SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received;  • the SUT shall send a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.								
SIP parameter	BYE:Reason header value	CV_	SIP, encapsi	ulated R	EL (	constructed from the ISDN			
values	RELEASE	<del></del>							
ISDN parameter values	RELEASE; cause value: (	PIXII	), location (	PIXII)					
Comments	ISDN		SUT			SIP-I			
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)			
	RELEASE	<b>→</b>							
	RELEASE COMPLETE	<b>←</b>							
					<del>(</del>	200 OK INVITE(ANM)			
				, i	<b>→</b>	ACK			
					<b>→</b>	BYE(REL)			
					<b>+</b>	200 OK BYE(RLC)			

TP605021	SIP reference: RF0	326	1 [4]		Q.1	ISDN/ISDN reference: 1912.5 [1], clause 7.7.1 2) 4)			
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection criteria	PICS 4/10								
ISDN selection criteria									
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message on receipt DISCONNECT message with <b>Cause value</b> CV_ISDN, <b>location</b> LOC_ISDN <b>before</b> a 200 OK response (any) message has been received which establishes a confirmed dialogue:  • the SUT shall hold the release procedure until a SIP 200 OK INVITE response has been received;  • the SUT shall send a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.								
SIP parameter values						constructed from the ISDN			
ISDN parameter values	DISCONNECT; cause val	lue: (	PIXIT), locat	ion (Pl	XIT)				
Comments	ISDN		SUT			SIP-I			
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ALERTING	<del>(</del>			+	180 Ringing(ACM			
	CONNECT	+			<b>←</b>	200 OK INVITE(ANM)			
					<b>→</b>	ACK			
			Conversa	tion					
	DISCONNECT	<b>→</b>			<b>→</b>	BYE(REL)			
	RELEASE	+		_	+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>							

TP605022	SIP reference: RF0	C 3261 [4]		Q.1	ISDN/ISDN reference: 912.5 [1], clause 7.7.1 2) 3				
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection	PICS 4/10								
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message with Cause value CV_ISDN, location LOC_ISDN before a 200 OK response message has been received:  • the SUT shall hold the REL message a CANCEL is sent when any SIP response was been received;  • on subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.								
SIP parameter	BYE:Reason header value			ted REL	constructed from the ISDN				
values ISDN parameter values	RELEASE COMPLETE; c	ause value	: (PIXIT),	location	(PIXIT)				
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
				+	100 Trying				
	RELEASE COMPLETE	<b>→</b>							
				→	CANCEL(REL)				
				<b>←</b>	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
					200 OK CANCEL				
				<b>→</b>	BYE(REL)				
				+	200 OK BYE(RLC)				

TP605023	SIP reference: RF0	C 3261 [4]	Q.	ISDN/ISDN reference: 1912.5 [1], clause 7.7.1 2) 3					
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT								
SIP selection	PICS 4/10								
criteria									
ISDN selection criteria									
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE message with Cause value CV_ISDN, location LOC_ISDN message before a 200 OK response message has been received:  • the SUT shall hold the release procedure. A CANCEL is sent when any SIP response was been received;  • on subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.								
SIP parameter		e CV_SIP, encaps	ulated REL	constructed from the ISDN					
values	RELEASE								
ISDN parameter values	RELEASE; cause value:	(PIXIT), <b>location</b> (	(PIXIT)						
Comments	ISDN	SUT		SIP-I					
	SETUP	<b>→</b>	→	INVITE(IAM)					
			+	100 Trying					
	RELEASE	<b>→</b>							
	RELEASE COMPLETE	<del>-</del>	→	CANCEL(REL)					
			+	200 OK INVITE(ANM)					
			→	ACK					
			+	200 OK CANCEL					
			→	BYE(REL)					
		<b>→</b>	+	200 OK BYE(RLC)					

TP605024	SIP reference: RF	C 3261 [4]		Q.	ISDN/ISDN reference: 1912.5 [1], clause 7.7.1 2) 3	
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT					
SIP selection criteria	PICS 4/10					
ISDN selection criteria						
Test purpose  SIP parameter	<ul> <li>Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message with Cause value CV_ISDN, location LOC_ISDN before a 200 OK response message has been received: <ul> <li>the SUT shall hold the release procedure. A CANCEL is sent when any SIP response was 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. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.</li> </ul> </li> <li>BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN</li> </ul>					
values	DISCONNECT	,	ooapou			
ISDN parameter values	DISCONNECT; cause va	lue: (PIXIT	), location	n (PIXIT)		
Comments	ISDN		SUT		SIP-I	
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)	
				<del>(</del>	100 Trying	
	DISCONNECT	→				
	RELEASE ← CANCEL(REL)					
	RELEASE COMPLETE	→		<del>(</del>	200 OK INVITE(ANM)	
				→	ACK	
				+	200 OK CANCEL	
				<b>→</b>	BYE(REL)	
				+	200 OK BYE(RLC)	

TP605025	SIP reference: RF	C 3261	[4]	Q	ISDN/ISDN reference: .1912.5 [1], clause 7.7.1 3)
TSS reference	ISDN-SIP /Basic call/Rec	eipt of	RELEASE		:
SIP selection criteria	PICS 4/10				
ISDN selection criteria					
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message with Cause value CV_ISDN, location LOC_ISDN before an early dialogue with the message defined as SIP_MESSAGE has been established:  • the SUT shall hold the release procedure until a SIP_MESSAGE_VA response has been received;  • the SUT shall send a CANCEL request or a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.				
SIP parameter values	BYE:Reason header value RELEASE COMPLETE	e CV_S	SIP, encaps	ulated REL	constructed from the ISDN
ISDN parameter values	RELEASE COMPLETE; c	ause v	<b>/alue:</b> (PIXI	T), location	(PIXIT)
Comments	ISDN		SUT		SIP-I
	SETUP	<b>→</b>		→	INVITE(IAM)
				+	100 Trying
	RELEASE COMPLETE	<b>→</b>			
				+	SIP_MESSAGE_VA
	CASE A				
				→	CANCEL(REL)
				+	200 OK CANCEL
				<b>←</b>	487 Request Terminated
				<b>→</b>	ACK
	CASE B				
				→	BYE(REL)
				+	200 OK BYE(RLC)
				+	487 Request Terminated
				<b>→</b>	ACK

TP605026	SIP reference: RF	C 326	1 [4]	Q	ISDN/ISDN reference: .1912.5 [1], clause 7.7.1 3)		
TSS reference	ISDN-SIP /Basic call/Rec	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT					
SIP selection	PICS 4/10	PICS 4/10					
criteria							
ISDN selection							
criteria							
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE message with Cause value CV_ISDN, location LOC_ISDN before an early dialogue with the message defined as SIP_MESSAGE has been established:  • the SUT shall hold the release procedure until a SIP_MESSAGE_VA response has been received;  • the SUT shall send a CANCEL request or a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.						
SIP parameter		BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN					
values	RELEASE		,				
ISDN parameter	RELEASE; cause value:	(PIXIT	), location (	PIXIT)			
values		`		,			
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		→	INVITE(IAM)		
				+	100 Trying		
	RELEASE	<b>→</b>					
	RELEASE COMPLETE	<b>←</b>					
				+	SIP_MESSAGE_VA		
	CASE A						
				→	CANCEL(REL)		
				+	200 OK CANCEL		
				+	487 Request Terminated		
				<b>→</b>	ACK		
	CASE B						
				<b>→</b>	BYE(REL)		
				+	200 OK BYE(RLC)		
				+	487 Request Terminated		
1		_			ACK		

TP605027	SIP reference: RF	C 326	1 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)		
TSS reference	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT					
SIP selection criteria	PICS 4/10					
ISDN selection criteria						
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message with Cause value CV_ISDN, location LOC_ISDN before an early dialogue with the message defined as SIP_MESSAGE has been established:  • the SUT shall hold the release procedure until a SIP_MESSAGE_VA response has been received;  • the SUT shall send a CANCEL request or a BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.					
SIP parameter values	BYE:Reason header valu DISCONNECT	e CV_	SIP, encaps	ulated REL	constructed from the ISDN	
ISDN parameter values	DISCONNECT; cause va	lue: (	PIXIT), <b>locat</b>	ion (PIXIT)		
Comments	ISDN		SUT		SIP-I	
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)	
				+	100 Trying	
	DISCONNECT	<b>→</b>			, ,	
	RELEASE	+				
	RELEASE COMPLETE	<b>→</b>				
				+	SIP_MESSAGE_VA	
	CASE A					
				<b>→</b>	CANCEL(REL)	
		1		<del>-</del>	200 OK CANCEL	
				+	487 Request Terminated	
				<b>→</b>	ACK	
	CASE B				1.19.1	
	_			<b>→</b>	BYE(REL)	
		1		<del>-</del>	200 OK BYE(RLC)	
				+	487 Request Terminated	
				<b>→</b>	ACK	
			l		7.01	

TP605028	SIP reference: RF0	C 3261 [4]		Q	ISDN/ISDN reference: .1912.5 [1], clause 7.7.1 3)	
TSS reference	ISDN-SIP /Basic call/Rece	eipt_of_RELEA	SE_or_DIS	COI	NNECT	
SIP selection	PICS 4/10					
criteria						
ISDN selection						
criteria						
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE COMPLETE message with Cause value CV_ISDN, location LOC_ISDN after a 200 OK response message has been received:  • the SUT shall send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.					
SIP parameter		e CV_SIP, enca	psulated F	REL	constructed from the ISDN	
values	RELEASE COMPLETE				(51) (15)	
ISDN parameter values	RELEASE COMPLETE; c	ause value: (F	IXII), <b>Ioc</b> a	ition	(PIXII)	
Comments	ISDN	SI	JT		SIP-I	
	SETUP	→		<b>→</b>	INVITE(IAM)	
	ALERTING	+		<b>←</b>	180 Ringing(ACM	
	CONNECT	<del>-</del>		<b>←</b>	200 OK INVITE(ANM)	
	RELEASE COMPLETE	<b>→</b>				
				<b>→</b>	ACK	
				<b>→</b>	BYE(REL)	
				+	200 OK BYE(RLC)	

TP605029	SIP reference: RF0	C 3261 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.1 3)		
TSS reference	ISDN-SIP /Basic call/Rece	ipt_of_RELE	ASE_or_DI	SCO	NNECT	
SIP selection	PICS 4/10					
criteria						
ISDN selection criteria						
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE message with Cause value CV_ISDN, location LOC ISDN after a 200 OK response message has been received:  • the SUT shall send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.					
SIP parameter values	BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN RELEASE					
ISDN parameter values	RELEASE; cause value: (	(PIXIT), <b>loca</b>	tion (PIXIT)			
Comments	ISDN		SUT		SIP-I	
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)	
	ALERTING	+		+	180 Ringing(ACM	
	CONNECT	+	· · · · · · · · · · · · · · · · · · ·	+	200 OK INVITE(ANM)	
	RELEASE	<b>→</b>	<u> </u>			
	RELEASE COMPLETE	<del>-</del>		<b>→</b>	ACK	
				<b>→</b>	BYE(REL)	
				<b>←</b>	200 OK BYE(RLC)	

TP605030	SIP reference: RFC 3261 [4]				Q.	ISDN/ISDN reference: 1912.5 [1], clause 7.7.1 3)
TSS reference	ISDN-SIP /Basic call/Rece	ipt_o	f_RELEASE_	or_DIS	CON	NNECT
SIP selection criteria	PICS 4/10					
ISDN selection criteria						
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message with Cause value CV_ISDN, location LOC_ISDN after a 200 OK response message has been received:  • the SUT shall send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.					
SIP parameter values	BYE:Reason header value DISCONNECT	CV_	SIP, encapsi	ulated F	REL	constructed from the ISDN
ISDN parameter values	DISCONNECT; cause val	ue: (	PIXIT), locat	ion (PI)	XIT)	
Comments	ISDN		SUT			SIP-I
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)
	ALERTING	4			+	180 Ringing(ACM
	CONNECT	4			+	200 OK INVITE(ANM)
	DISCONNECT	<b>→</b>				
	RELEASE	+			<b>→</b>	ACK
	RELEASE COMPLETE	<b>^</b>			<b>→</b>	BYE(REL)
					<b>←</b>	200 OK BYE(RLC)

TP605031	SIP reference: RF	C 326	1 [4]		ISDN/ISDN reference:		
			. [.]	G	0.1912.5 [1], clause 7.7.1 3)		
TSS reference	ISDN-SIP /Basic call/Rece	ISDN-SIP /Basic call/Receipt_of_RELEASE_or_DISCONNECT					
SIP selection	PICS 4/10	•					
criteria							
ISDN selection							
criteria							
Test purpose	Ensure that the SUT after	recei	ving the SET	UP with the	complete called party number,		
					COMPLETE message with Cause		
					alogue with the SIP message defined		
	with the SIP_MESSAGE_						
					e cause Value Indicator parameter		
CID navamatar	defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP;						
SIP parameter values	RELEASE COMPLETE	BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN					
ISDN parameter	RELEASE COMPLETE; c	21160	value: (DIXI	T\ location	(DIVIT)		
values	INCLEASE COMIT LETE, C	ause	value. (FIXI	i), iocatioi	I (I IXII)		
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	SIP_MESSAGE_VA		
	RELEASE COMPLETE	<b>→</b>					
	CASE A						
				→	CANCEL(REL)		
				+	200 OK CANCEL		
				+	487 Request Terminated		
	→ ACK						
	CASE B						
				<b>→</b>	BYE(REL)		
				+	200 OK BYE(RLC)		
				+	487 Request Terminated		
				<b>→</b>	ACK		

TP605032	SIP reference: RF	C 3261 [4]	Q	ISDN/ISDN reference: .1912.5 [1], clause 7.7.1 3)		
TSS reference	ISDN-SIP /Basic call/Rec	eipt_of_RELEASE	_or_DISCOI	NNECT		
SIP selection	PICS 4/10					
criteria						
ISDN selection criteria						
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a RELEASE message with Cause value CV_ISDN, location LOC_ISDN after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established:  • the SUT shall send a CANCEL or BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP;					
SIP parameter	BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN					
values	RELEASE					
ISDN parameter	RELEASE; cause value:	(PIXIT), location	(PIXIT)			
values						
Comments	ISDN	SUT		SIP-I		
	SETUP	→	<b>→</b>	INVITE(IAM)		
			<b>←</b>	SIP_MESSAGE_VA		
	RELEASE	→				
	RELEASE COMPLETE	<del>-</del>				
	CASE A					
			<b>→</b>	CANCEL(REL)		
			<b>←</b>	200 OK CANCEL		
			+	487 Request Terminated		
			<b>→</b>	ACK		
	CASE B					
			<b>→</b>	BYE(REL)		
			+	200 OK BYE(RLC)		
			+	487 Request Terminated		
			→	ACK		

TP605032	SIP reference: RF	C 3261 [4]	0	ISDN/ISDN reference: .1912.5 [1], clause 7.7.1 3)	
TSS reference	ISDN-SIP /Basic call/Rec	eint of RELEASE		/	
SIP selection	PICS 4/10	elpt_ol_INELEAGE	_01_D10001	NINEOI	
criteria	1 100 4/10				
ISDN selection criteria					
	Francis de at the OUT after	OFT	TIDide de e	letelledteresconter-	
Test purpose	Ensure that the SUT after receiving the SETUP with the complete called party number, sending an INVITE message. On receipt of a DISCONNECT message with Cause value CV_ISDN, location LOC_ISDN after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established:  • the SUT shall send a CANCEL or BYE request. The cause Value Indicator parameter defined as CV_ISDN shall be mapped to the Reason header field defined as CV_SIP.				
SIP parameter values	BYE:Reason header value CV_SIP, encapsulated REL constructed from the ISDN DISCONNECT				
ISDN parameter values	DISCONNECT; cause va	llue: (PIXIT), locat	ion (PIXIT)		
Comments	ISDN	SUT		SIP-I	
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)	
			+	SIP_MESSAGE_VA	
	DISCONNECT	<b>→</b>			
	RELEASE	<del>-</del>			
	RELEASE COMPLETE	<b>→</b>			
	CASE A				
			→	CANCEL(REL)	
			+	200 OK CANCEL	
			←	487 Request Terminated	
			<b>→</b>	ACK	
	CASE B				
			<b>→</b>	BYE(REL)	
			<del>-</del>	200 OK BYE(RLC)	
			+	487 Request Terminated	
			→	ACK	

#### Table 29

Values fo	TP605010, TP605011, TP605012; TP605016, TP605017, TP605018 TP605025; TP605026; TP605027; TP605031; TP605032 TP605033	
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 30

	Values for test purposes 306021 - 306033				
←SIP Mess	age	← REL			
Reason hea	nder field	Cause Indicators parameter			
CV_SIP		CV_ISDN			
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 31: 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
parameter neid	ileiu	neader neid	
-	-	Protocol	"Q.850"
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 [3] (see note 2)

NOTE 1: "XX" is the Cause Value as defined in ITU-T Recommendation Q.850 [3].

NOTE 2: Due to the fact that the Cause Indications parameter does not include the definition text as defined in table 1/ITU-T Recommendation Q.850 [3] this is based on provisioning in the O-IWU.

#### A.1.1.1.2.6 Receipt of a backward final response or BYE Message

TP606001	SIP reference: RFC	3261 [	4]		SDN/ISDN reference: 1912.5 [1], clause 7.7.2					
TSS reference	ISDN-SIP /Basic call/Rece	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE								
SIP selection										
criteria										
ISDN selection criteria										
Test purpose	receipt of a BYE message included:  sends a DISCONNEC									
SIP parameter values	BYE: ISUP REL encapsul	ated in	the MIME	body						
ISDN parameter values	DISCONNECT: Cause va	lue cons	structed f	rom the encap	sulated REL					
Comments	ISDN		SI	JT	SIP-I					
	SETUP	<b>→</b>		→	INVITE(IAM)					
	ALERTING	+		+	180 Ringing(ACM					
	CONNECT	+		+	200 OK INVITE(ANM)					
				→	ACK					
			Conve	rsation						
	DISCONNECT	+		+	BYE(REL)					
	RELEASE	<b>→</b>		→	200 OK BYE(RLC)					
	RELEASE COMPLETE	+								

TP606002	SIP reference: RFC	3261 [	4]			DN/ISDN reference: 912.5 [1], clause 7.7.2			
TSS reference	ISDN-SIP /Basic call/Rec	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection criteria									
ISDN selection criteria									
Test purpose	receipt of a BYE message included:  sends a DISCONNEC	Ensure that the SUT after receiving the SETUP sends out a INVITE message and on receipt of a BYE message where a Reason header field with Q.850 Cause Value is included:  • sends a DISCONNECT message constructed from the encapsulated REL in the received BYE MIME body.							
SIP parameter values	BYE: ISUP REL encapsul	ated in	the MIME	body, R	eason	header value = (PIXIT)			
ISDN parameter values	DISCONNECT: Cause va	lue con	structed f	rom the e	ncaps	sulated REL			
Comments	ISDN		SI	JT		SIP-I			
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)			
	ALERTING	+			+	180 Ringing(ACM			
	CONNECT	+			+	200 OK INVITE(ANM)			
					<b>→</b>	ACK			
			Conve	rsation					
	DISCONNECT	+			+	BYE(REL)			
	RELEASE	<b>→</b>			<b>→</b>	200 OK BYE(RLC)			
	RELEASE COMPLETE	+		•					

Table 32: Mapping of SIP Reason header fields into Cause Indicators parameter

component of SIP Reason header field	I.E. value		
Protocol	"Q.850"	Cause Indication parameter	-
protocol-cause	"cause = XX" (see note 1)	Cause Value	constructed from the encapsulated REL
-	-	Location	constructed from the encapsulated REL

TP606003	SIP reference: RFC 3	261 [4]			SDN/ISDN reference: 912.5 [1], clause 7.7.6			
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N2, before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is not included defined as SIP_Failure_VA:  • sends a DISCONNECT or RELEASE message with the Cause value set to the value of the encapsulated REL.							
SIP parameter values	SIP_Failure_VA:ISUP REL	encapsu	ated in the MI	IME bo	dy			
ISDN parameter values	DISCONNECT/RELEASE: c	ause va	lue: mapped	from th	ne encapsulated REL			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		→	INVITE(IAM)			
	SETUP ACK	+			, , ,			
	CASE A			+	SIP_Failure_VA(REL)			
	RELEASE	+		<b>→</b>	ACK			
	RELEASE COMPLETE →							
	CASE B							
	DISCONNECT	+						
	RELEASE	<b>→</b>						
	RELEASE COMPLETE	+						

TP606004	SIP reference: RFC 3	SIP reference: RFC 3261 [4]			SDN/ISDN reference: 912.5 [1], clause 7.7.6			
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection								
criteria								
ISDN selection criteria								
Test purpose	that the SUT in state N3, be Failure message (4xx, 5xx, <b>not</b> included defined as SIP	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N3, before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is not included defined as SIP_Failure_VA:  • sends a DISCONNECT or RELEASE message with the Cause value set to the value of the encapsulated REI.						
SIP parameter values	SIP_Failure_VA: ISUP REL	encaps	ulated in the M	IIME bo	ody			
ISDN parameter values	DISCONNECT/RELEASE: (	cause v	alue: mapped	from th	ne encapsulated REL			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		→	INVITE(IAM)			
	CALL PROCEEDING	+						
	CASE A			+	SIP_Failure_VA(REL)			
	RELEASE ← → ACK							
	RELEASE COMPLETE	→						
	CASE B							
	DISCONNECT	+						
	RELEASE	<b>→</b>						
	RELEASE COMPLETE	+						

TP606005	SIP reference: RFC	3261 [4]			SDN/ISDN reference: 912.5 [1], clause 7.7.6			
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection criteria								
ISDN selection criteria								
Test purpose  SIP parameter	Ensure that the SUT after receiving the SETUP 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 out an INVITE message. Ensure that the SUT in state N3 on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is not included defined as SIP_Failure_VA;  • sends a DISCONNECT or RELEASE message with the Cause value set to the value of the encapsulated REL.  SIP_Failure_VA: no ISUP REL encapsulated in the MIME body							
values			, a.a.c. a		. 200,			
ISDN parameter values	DISCONNECT/RELEASE:	cause valu	ie: mapped	from th	e encapsulated REL			
Comments	ISDN		SUT		SIP-I			
	SETUP	→		<b>→</b>	INVITE(IAM)			
	CALL PROCEEDING	+						
	CASE A			+	SIP_Failure_VA(REL)			
	DISCONNECT	+		<b>→</b>	ACK			
	RELEASE	<b>→</b>						
	RELEASE COMPLETE	+						
	CASE B							
	RELEASE	+						
	RELEASE COMPLETE	<b>→</b>						

Table 33

	Values for test purpose TP606003, TP606004, TP606005						
VA	←REL (Cause Value) CV_ ISDN	←4XX/5XX/6XX SIP message SIP_Failure_VA					
VA_01	127 Interworking	400 Bad Request					
VA_02	127 Interworking	402 Payment Required					
VA_03	127 Interworking	403 Forbidden					
VA_04	1 Unallocated number	404 Not Found					
VA_05	127 Interworking	405 Method Not Allowed					
VA_06	127 Interworking	406 Not Acceptable					
VA_07	127 Interworking	408 Request Timeout					
VA_08	22 Number changed (without diagnostic)	410 Gone					
VA_9	127 Interworking	423 Interval Too Brief					
VA_10	20 Subscriber absent	480 Temporarily Unavailable					
VA_11	127 Interworking	481 Call/Transaction does not exist					
VA_12	127 Interworking	482 Loop Detected					
VA_13	127 Interworking	483 Too many hops					
VA_14	127 Interworking	485 Ambiguous					
VA_15	17 User busy	486 Busy Here					
VA_16	127 Interworking	488 Not acceptable here					
VA_17	127 Interworking	493 Undecipherable					
VA_18	127 Interworking	500 Server Internal error					
VA_19	127 Interworking	501 Not implemented					
VA_20	127 Interworking	502 Bad Gateway					
VA_21	127 Interworking	504 Server timeout					
VA_22	17 User busy	600 Busy Everywhere					
VA_23	21 Call rejected	603 Decline					
VA_24	1 Unallocated number	604 Does not exist anywhere					
VA_25	127 Interworking	606 Not acceptable					

TP606006	SIP reference: RFC 32	61 [4]				SDN/ISDN reference: 912.5 [1], clause 7.7.6		
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection	PICS 4/12							
criteria								
ISDN selection criteria								
Test purpose	SETUP message sends out a having received an backward	Ensure that the SUT if the SIP Failure response is interworked to ISDN after receiving an SETUP message sends out an INVITE message. Ensure that the SUT in state N2, before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is <b>not</b> included defined as						
	SIP_Failure_VA:	WILII G	2.000 Ca	ause va	liue is i	not included defined as		
	<ul> <li>sends a DISCONNECT of ISDN.</li> </ul>	r REL	.EASE r	nessag	e with	the Cause value set to CV_		
SIP parameter								
values								
ISDN parameter values	DISCONNECT/RELEASE; ca	use v	alue: C	V_ISDI	N (PIXI	T)		
Comments	ISDN		S	UT		SIP-I		
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)		
	SETUP ACK	+						
	CASE A				+	SIP_Failure_VA		
	RELEASE	+			<b>→</b>	ACK		
	RELEASE COMPLETE	<b>→</b>						
	CASE B							
	DISCONNECT	<b>←</b>						
	RELEASE	<b></b>						
	RELEASE COMPLETE	+						

TP606007	SIP reference: RFC 32	61 [4]			SDN/ISDN reference: 912.5 [1], clause 7.7.6			
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection	PICS 4/12							
criteria								
ISDN selection								
criteria								
Test purpose	Ensure that the SUT if the SIP Failure response is interworked to ISDN after receiving an SETUP message sends out an INVITE message. Ensure that the SUT in state N3, before having received an backward message, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is <b>not</b> included defined as SIP_Failure_VA:  sends a DISCONNECT or RELEASE message with the Cause value set to CV_ISDN.							
SIP parameter								
values								
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CALL PROCEEDING	+						
	CASE A			+	SIP_Failure_VA			
	RELEASE	<b>←</b>		<b>→</b>	ACK			
	RELEASE COMPLETE	<b>→</b>						
	CASE B							
	DISCONNECT	+						
	RELEASE	<b>→</b>	·					
	RELEASE COMPLETE	+						

Table 34

	Values for test purposes TP606006, TP606007						
VA	←REL (Cause Value) CV_ ISDN	←4XX/5XX/6XX SIP message SIP_Failure_VA					
VA_01	127 Interworking	401 Unauthorized					
VA_02	127 Interworking	407 Proxy authentication required					
VA_03	127 Interworking	413 Request Entity too long					
VA_04	127 Interworking	414 Request-uri too long					
VA_05	127 Interworking	415 Unsupported Media type					
VA_06	127 Interworking	416 Unsupported URI scheme					
VA_07	127 Interworking	420 Bad Extension					
VA_08	127 Interworking	421 Extension required					
VA_09	127 Interworking	503 Service Unavailable					
VA_10	127 Interworking	505 Version not supported					
VA_11	127 Interworking	513 Message too large					
VA_12	127 Interworking	580 Precondition failure					

TP606008	SIP reference: RF	C 32	61 [4]		(	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6				
TSS reference	ISDN-SIP /Basic call/Red	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE								
SIP selection criteria	NOT PICS 4/12	NOT PICS 4/12								
ISDN selection criteria										
Test purpose	of a Failure message (4x is <b>not</b> included defined a	Ensure that the SUT after receiving the SETUP sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value is <b>not</b> included defined as SIP_Failure_VA:  • no action is taken on the ISDN.								
SIP parameter values										
ISDN parameter values										
Comments	ISDN		SUT			SIP-I				
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)				
					+	SIP_Failure_VA				
					<b>→</b>	ACK				
		Further SIP procedures apply								

Table 35

V	/alues for test purposes TP606008
VA	←4XX/5XX/6XX SIP message
	SIP_Failure_VA
VA_01	401 Unauthorized
VA_02	407 Proxy authentication required
VA_03	413 Request Entity too long
VA_04	414 Request-uri too long
VA_05	415 Unsupported Media type
VA_06	416 Unsupported URI scheme
VA_07	420 Bad Extension
VA_08	421 Extension required
VA_09	503 Service Unavailable
VA_10	505 Version not supported
VA_11	513 Message too large
VA_12	580 Precondition failure

TP606009	SIP reference: RFC	3261 [	4]			DN/ISDN reference: 912.5 [1], clause 7.7.6	
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection criteria							
ISDN selection criteria							
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message, on receipt of a Failure message <b>487 Request terminated</b> where a Reason header field with Q.850 Cause Value is <b>not</b> included:  • no action is taken on the ISDN if a CANCEL request was previously sent before answer to an INVITE.						
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SI	JT		SIP-I	
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)	
					+	100 Trying	
	RELEASE	<b>→</b>			<b>→</b>	CANCEL(REL)	
	RELEASE COMPLETE	+			+	200 OK CANCEL	
				•	+	487 Request terminated	
					<b>→</b>	ACK	

TP606010	SIP reference: RFC 3	261 [	4]		DN/ISDN reference: 912.5 [1], clause 7.7.6		
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection criteria							
ISDN selection criteria							
Test purpose	of a Failure message 491 R	Ensure that the SUT after receiving the SETUP sends out an INVITE message, on receipt of a Failure message <b>491 Request Pending</b> where a Reason header field with Q.850 Cause Value is <b>not</b> included:  • no action is taken on the ISDN					
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
				+	100 Trying		
				+	491 Request Pending		
				<b>→</b>	ACK		

TP606011	SIP reference: RFC 32	61 [4]				DN/ISDN reference: 912.5 [1], clause 7.7.6
TSS reference	ISDN-SIP /Basic call/Receipt_	_of_ba	ackward	d_final_r	espons	se_or_BYE
SIP selection criteria	NOT PICS 4/11					
ISDN selection criteria						
Test purpose	Ensure that the SUT after receiving the SETUP 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 Q.850 Cause Value is <b>not</b> included:  • sends a RELEASE or DISCONNECT message with the Cause value set to CV_ISDN.					
SIP parameter values						
ISDN parameter values	RELEASE/DISCONNECT; ca	use v	<b>alue</b> : C	V_ISDN	1	
Comments	ISDN		S	UT		SIP-I
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)
					+	SIP MESSAGE_VA
	CASE A				+	SIP_Failure_VA
	RELEASE	RELEASE ← → ACK				
	RELEASE COMPLETE →					
	CASE B					
	DISCONNECT	+				
	RELEASE	<b>→</b>	·			
	RELEASE COMPLETE	+				

Table 36

	Values for test purpose TP606011			
VA	SIP MESSAGE_VA			
VA_1	181 Call Is Being Forwarded			
VA_2	182 Queued			
VA_3	183 Session Progress			

TP606012	SIP reference: RFC 3261 [4]			ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6				
TSS reference	ISDN-SIP /Basic call/Receip	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection								
criteria								
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message a 180 ringing message has been received on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA where a Reason header field with Q.850 Cause Value is not included:  • sends a DISCONNECT message with the Cause value CV_ ISDN.							
SIP parameter	SIP_Failure_VA							
values								
ISDN parameter	RELEASE/DISCONNECT; c	ause v	alue: CV_ISD	N				
values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	DISCONNECT	+		+	SIP_Failure_VA			
	RELEASE	<b>→</b>		<b>→</b>	ACK			
	RELEASE COMPLETE	+						

Table 37

	Values for test purposes TP606012						
VA	←REL (Cause Value) CV_ ISDN	←4XX/5XX/6XX SIP message SIP_Failure_VA					
VA_01	127 Interworking	408 Request timeout					
VA_02	17 User busy	486 Busy Here					
VA_03	17 User busy	600 Busy Everywhere					
VA_04	21 Call rejected	603 Decline					

TP606013	SIP reference: RFC	3261 [4]			SDN/ISDN reference: 1912.5 [1], clause 7.7.6			
TSS reference	ISDN-SIP /Basic call/Recei	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection criteria	PICS 4/11							
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N2, before having received an backward 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 Q.850 Cause Value is included:  • sends a DISC message. The Cause Value in the header field set to CV_SIP is mapped to the ISDN Cause Value field in the ISDN REL message with the Cause value set to CV_ ISDN.							
SIP parameter values	SIP_Failure_VA Reason he	eader CV_	SIP (PIXIT)					
ISDN parameter values	DISCONNECT/RELEASE	cause valu	ie CV_ ISDN	(PIXIT	)			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	SETUP ACK	+						
				+	SIP MESSAGE_VA			
	CASE A			+	SIP_Failure_VA			
	RELEASE	+		<b>→</b>	ACK			
	RELEASE COMPLETE	<b>→</b>	· · · · · · · · · · · · · · · · · · ·					
	CASE B							
	DISCONNECT	<b>←</b>						
	RELEASE	→						
	RELEASE COMPLETE	<b>←</b>						

TP606014	SIP reference: RFC	3261 [4]			SDN/ISDN reference: 912.5 [1], clause 7.7.6		
TSS reference	ISDN-SIP /Basic call/Recei	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE					
SIP selection	PICS 4/11						
criteria							
ISDN selection criteria							
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N3, before having received an backward 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 Q.850 Cause Value is included:  • sends a REL message. The Cause Value in the header field set to CV_SIP is mapped to the ISDN Cause Value field in the ISDN REL message with the Cause value set to CV_ISDN.						
SIP parameter values	SIP_Failure_VA Reason he	ader CV	_ SIP (PIXIT)				
ISDN parameter values	DISCONNECT/RELEASE of	cause val	ue CV_ ISDN	(PIXIT			
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	CALL PROCEEDING	+					
				+	SIP MESSAGE_VA		
	CASE A			<b>←</b>	SIP_Failure_VA		
	RELEASE	+		<b>→</b>	ACK		
	RELEASE COMPLETE	<b>→</b>					
	CASE B						
	DISCONNECT	+					
	RELEASE	<b>→</b>					
	RELEASE COMPLETE	+					

TP606015	SIP reference: RFC 326	61 [4]			DN/ISDN reference: 912.5 [1], clause 7.7.6		
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE						
SIP selection	PICS 4/11						
criteria							
ISDN selection criteria							
Test purpose	<ul> <li>Ensure that the SUT after receiving the SETUP 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:         <ul> <li>sends out an INVITE message. Ensure that the SUT in state N3, having received an backward message indicating that sufficient number of digits has been received to route the call to the called party, on receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with Q.850 Cause Value Cause Value is included;</li> <li>sends a REL message. The Cause Value in the header field set to C V_SIP is mapped to the ISDN Cause Value field in the ISDN REL message with the Cause value set to CV_ISDN.</li> </ul> </li> </ul>						
SIP parameter values	SIP_Failure_VA Reason head	er CV_ SI	P (PIXIT)				
ISDN parameter values	DISCONNECT/RELEASE cau	ise value C	V_ ISDN (F	PIXIT			
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	CALL PROCEEDING	<b>←</b>					
				<b>←</b>	SIP MESSAGE_VA		
	CASE A			<b>←</b>	SIP_Failure_VA		
	RELEASE	<b>←</b>		<b>→</b>	ACK		
	RELEASE COMPLETE	<b>→</b>					
	CASE B						
	DISCONNECT	+					
	RELEASE	<b>→</b>					
	RELEASE COMPLETE	<b>←</b>					

Table 38

	Values for test purpose TP606013, 606014, 606015				
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 39

	Values for test purposes TP606011, TP606013, TP606014, TP606015						
VA	←REL (Cause Value)	←4XX/5XX/6XX SIP message					
	CV_ISDN	SIP_Failure_VA					
VA_01	127 Interworking	400 Bad Request					
VA_02	127 Interworking	402 Payment Required					
VA_03	127 Interworking	403 Forbidden					
VA_04	1 Unallocated number	404 Not Found					
VA_05	127 Interworking	405 Method Not Allowed					
VA_06	127 Interworking	406 Not Acceptable					
VA_07	127 Interworking	408 Request Timeout					
VA_08	22 Number changed (without diagnostic)	410 Gone					
VA_09	127 Interworking	423 Interval Too Brief					
VA_10	20 Subscriber absent	480 Temporarily Unavailable					
VA_11	127 Interworking	481 Call/Transaction does not exist					
VA_12	127 Interworking	482 Loop Detected					
VA_13	127 Interworking	483 Too many hops					
VA_14	127 Interworking	485 Ambiguous					
VA_15	17 User busy	486 Busy Here					
VA_16	127 Interworking	488 Not acceptable here					
VA_17	No mapping.	491 Request Pending					
VA_18	127 Interworking	493 Undecipherable					
VA_19	127 Interworking	500 Server Internal error					
VA_20	127 Interworking	501 Not implemented					
VA_21	127 Interworking	502 Bad Gateway					
VA_22	127 Interworking	504 Server timeout					
VA_23	17 User busy	600 Busy Everywhere					
VA_24	21 Call rejected	603 Decline					
VA_25	1 Unallocated number	604 Does not exist anywhere					
VA_26	127 Interworking	606 Not acceptable					

TP606016	SIP reference: RFC 32	261 [4]	ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6					
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection criteria	NOT PICS 4/17							
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N2, before having received an backward message, on receipt of a response message (3xx) defined as SIP_Response_VA, the SUT:  • sends a DISC message with the Cause value 127 Interworking.							
SIP parameter values				-				
ISDN parameter values	REL; cause value: CV_ISDN	N						
Comments	ISDN		SUT	SIP-I				
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)				
	SETUP ACK	<del>-</del>		, , ,				
	CASE A		+	SIP_Response_VA				
	RELEASE	+	→	ACK				
	RELEASE COMPLETE	<b>→</b>						
	CASE B							
	DISCONNECT	+						
	RELEASE	<b>→</b>						
	RELEASE COMPLETE	<b>←</b>						

TP606017	SIP reference: RFC 3	3261 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6					
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE								
SIP selection	NOT PICS 4/17								
criteria									
ISDN selection criteria									
Test purpose	that the SUT in state N3, be response message (3xx) de	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N3, before having received an backward message, on receipt of a response message (3xx) defined as SIP_Response_VA, the SUT:  • sends a DISC message with the Cause value 127 Interworking.							
SIP parameter values									
ISDN parameter values	REL; cause value: CV_ISD	N							
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	CALL PROCEEDING	+							
	CASE A			+	SIP_Response_VA				
	RELEASE	+		<b>→</b>	ACK				
	RELEASE COMPLETE	<b>→</b>							
	CASE B								
	DISCONNECT	+							
	RELEASE	<b>→</b>							
	RELEASE COMPLETE	+							

TP606018	SIP reference: RFC 32	61 [4]	Q	ISDN/ISDN reference: .1912.5 [1], clause 7.7.6					
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE								
SIP selection	NOT PICS 4/17								
criteria									
ISDN selection									
criteria									
Test purpose  SIP parameter	Ensure that the SUT after receiving the SETUP message containing the complete <b>called</b> 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 out an INVITE message. Ensure that the SUT in state N3, on receipt of a response message (3xx) defined as SIP_Response_VA, the SUT;  sends a DISC message with the Cause value 127 Interworking.								
values									
ISDN parameter values	DISC; cause value: CV_ISDN	١							
Comments	ISDN		SUT	SIP-I					
	SETUP	<b>→</b>	-	INVITE(IAM)					
	CALL PROCEEDING	+							
	CASE A		€	SIP_Response_VA					
	RELEASE	+	7	→ ACK					
	RELEASE COMPLETE	<b>→</b>							
	CASE B								
	DISCONNECT	+							
	RELEASE	<b>→</b>							
	RELEASE COMPLETE	+							

TP606019	SIP reference: RFC				DN/ISDN reference: 912.5 [1], clause 7.7.6			
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE							
SIP selection criteria	PICS 4/17							
ISDN selection criteria								
Test purpose	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N2, before having received an backward message on receipt of a response message (3xx) defined as SIP_Response_VA, the SUT:  • sends an INVITE using the value of the Contact header field in the received SIP_Response_VA in the Request URI.							
SIP parameter	SIP_Response_VA Contact: URI of new destination							
values	INVITE: Request URI of new destination							
ISDN parameter values								
Comments	ISDN		SI	JT		SIP-I		
	SETUP	<b>→</b>			→	INVITE(IAM)		
	SETUP ACK	+						
					+	SIP_Response_VA		
					<b>→</b>	ACK		
					<b>→</b>	INVITE(IAM)		
	ALERTING	+			+	180 Ringing(ACM		
	CONNECT	+			+	200 OK INVITE(ANM)		
					<b>→</b>	ACK		
			Conve	rsation	•			
	DISCONNECT	+			+	BYE(REL)		
	RELEASE	<b>→</b>			<b>→</b>	200 OK BYE(RLC)		
	RELEASE COMPLETE	+						

TP606020	SIP reference: RFC 3261 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6						
TSS reference	ISDN-SIP /Basic call/Receipt_of_backward_final_response_or_BYE								
SIP selection criteria	PICS 4/17	PICS 4/17							
ISDN selection criteria									
Test purpose	that the SUT in state N3, response message (3xx)  sends an INVITE uside	Ensure that the SUT after receiving the SETUP sends out an INVITE message. Ensure that the SUT in state N3, before having received an backward message on receipt of a response message (3xx) defined as SIP_Response_VA, the SUT:  • sends an INVITE using the value of the Contact header field in the received SIP_Response_VA in the Request URI.							
SIP parameter	SIP_Response_VA Contact: URI of new destination								
values	INVITE: Request URI of new destination								
ISDN parameter									
values									
Comments	ISDN		SI	UT	SIP-I				
	SETUP	<b>→</b>		→	INVITE(IAM)				
	CALL PROCEEDING	+							
				+	SIP_Response_VA				
				→	ACK				
				<b>→</b>	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM				
	CONNECT	+		+	200 OK INVITE(ANM)				
				→	ACK				
			Conve	rsation					
	DISCONNECT	+		+	BYE(REL)				
	RELEASE	<b>→</b>		<b>→</b>	200 OK BYE(RLC)				
	RELEASE COMPLETE	+							

TP606021	SIP reference: RFC 3261 [4] ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6								
TSS reference	ISDN-SIP /Basic call/Rec	eipt_of_	backward	l_final_r					
SIP selection criteria	PICS 4/17								
ISDN selection criteria									
Test purpose	party number where the party number to indicate to call to the called party:  sends out an INVITE response message ( sends an INVITE usi	<ul> <li>sends out an INVITE message. Ensure that the SUT in state N3, on receipt of a response message (3xx) defined as SIP_Response_VA, the SUT;</li> </ul>							
SIP parameter values	SIP_Response_VA Conta	SIP_Response_VA Contact: URI of new destination INVITE: Request URI of new destination							
ISDN parameter values									
Comments	ISDN		SI	JT		SIP-I			
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)			
	CALL PROCEEDING	+							
					+	SIP_Response_VA			
					<b>→</b>	ACK			
					<b>→</b>	INVITE(IAM)			
	ALERTING	+			+	180 Ringing(ACM			
	CONNECT	+			+	200 OK INVITE(ANM)			
					<b>→</b>	ACK			
			Conve	rsation					
	DISCONNECT	+			+	BYE(REL)			
	RELEASE	<b>→</b>			<b>→</b>	200 OK BYE(RLC)			
	RELEASE COMPLETE	+				, ,			

Table 40

Values for test purpose TP606016, TP606017, TP606018 TP606019, TP606020, TP606021						
VA	VA SIP_Response_VA					
VA_1	300 Multiple Choices					
VA_2	301 Moved Permanently					
VA_3	302 Move Temporarily					
VA_4	305 Use Proxy					
VA 5	380 Alternative Service					

#### A.1.1.2.7 Autonomous release at the MGC

TP607001	SIP reference: RFC 3	3261 [4]		Q		N/ISDN reference: 2.5 [1], clause 7.7.6.1			
TSS reference	ISDN-SIP /Basic call/Autono	ISDN-SIP /Basic call/Autonomous_release							
SIP selection	PICS 3/2								
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT a 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 SUT is configured to propagate overlap signalling into the SIP network, the SUT:  • shall not clear immediately the bearer channel and shall instead start timer T <sub>OIW3</sub> . The RELEASE or DISCONNECT message shall only be sent if T <sub>OIW3</sub> expires.								
SIP parameter values									
ISDN parameter values									
Comments	ISDN		S	UT		SIP-I			
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)			
			Start tin	ner T <sub>OIW3</sub>	<b>←</b>	484 Address incomplete			
					<b>→</b>	ACK			
			Timeo	ut T <sub>OIW3</sub>					
	CASE A								
	RELEASE	<b>←</b>							
	RELEASE COMPLETE	→							
	CASE B								
	DISCONNECT	+							
	RELEASE	→							
	RELEASE COMPLETE	<b>←</b>							

TP607002	SIP reference: RFC 32	·61 [4]		ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.6.1					
TSS reference	ISDN-SIP /Basic call/Autonom	ous_release	<del></del>		[-],				
SIP selection criteria	NOT PICS 3/4								
ISDN selection criteria									
Test purpose	Ensure that the SUT a 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-IWU is not configured to propagate overlap signalling into the SIP network then the timer shall not be started and:  • the DISCONNECT message shall be sent immediately to the ISDN network.								
SIP parameter values				,					
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
				+	484 Address incomplete				
	CASE A			<b>→</b>	ACK				
	RELEASE	<b>←</b>							
	RELEASE COMPLETE	<b>→</b>							
	CASE B								
	DISCONNECT	<b>←</b>							
	RELEASE	<b>→</b>							
	RELEASE COMPLETE	+							

TP607003	SIP reference: RF	C 326		ISDN/ISDN reference: Q.1912.5 [1], clause 7.7.3					
TSS reference	ISDN-SIP /Basic call/Autonomous_release								
SIP selection	PICS 4/4 AND 4/5								
criteria									
ISDN selection									
criteria									
Test purpose	Ensure that the SUT when the internal resource reservation is unsuccessful and preconditions used, the SUT:  • sends a CANCEL or BYE to the SIP network;								
	<ul> <li>sends a RELEASE to</li> </ul>			•					
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		→	INVITE(IAM)				
					183 Session Progress				
					PRACK				
					200 OK PRACK				
			Internal reso	ource					
		res	ervation uns	uccessful					
	CASE A								
	RELEASE	<b>←</b>		<b>→</b>	CANCEL(REL)				
	RELEASE COMPLETE	<b>→</b>		<b>←</b>	200 OK CANCEL				
				←	487 Request Terminated				
				<b>→</b>	ACK				
	CASE B								
	RELEASE	+		<b>→</b>	BYE(REL)				
	RELEASE COMPLETE	<b>→</b>		+	200 OK BYE(RLC)				
				<del>(</del>	487 Request Terminated				
				→	ACK				

# A.1.1.2 Test purposes for ISDN/SIP Supplementary services

### A.1.1.2.1 Calling Line Identification Presentation (CLIP)

TP701001	SIP reference: RF0	C 3261 [4]		ISUP reference: Q.1912.5 [1],				
				Q.731	[i.2], clause 3.5.2.1.1			
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP		•					
SIP selection criteria								
ISUP selection criteria								
Test purpose		cessfully origir to "network pr	nate a call ha		calling party number with sentation restricted indicator			
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body							
ISDN parameter values	3 3 3,1	- 1 1	,	1	,			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		Cor	nmunication					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>	•					

TP701002	SIP reference: RFC 3	3261 [4]		ISUP reference: Q.1912.5 [1],				
			Q.	.731	[i.2], clause 3.5.2.1.1			
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Calling party number (network provided) with calling sub-address  Verify that the IUT can successfully originate a call having a calling party number with the screening indicator set to "network provided" and an access transport parameter containing the calling sub-address.							
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body							
ISDN parameter values								
Comments	ISDN	S	UT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		<del>(</del>	180 Ringing(ACM)			
	CONN	+		<del>(</del>	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		Commi	unication					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		<del>-</del>	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

TP701003	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 3.5.2.1.1				
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	Verify that the IUT can succes	Calling party number (user provided, verified and passed) Verify that the IUT can successfully originate a call having the calling party number with the screening indicator set to "user provided, verified and passed".						
SIP parameter	INVITE: Content-Type: applic	ation/IS	SUP; IAM 6	encapsula	ated ir	n the MIME body		
values	180 Ringing: Content-Type: a	pplicati	ion/ISUP;	ACM enc	apsul	ated in the MIME body		
ISDN parameter values								
Comments	ISDN		SU <sup>*</sup>	Τ		SIP-I		
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)		
	ALERTING	+			+	180 Ringing(ACM)		
	CONN	+			+	200 OK INVITE(ANM)		
					<b>→</b>	ACK		
	Communication							
	DISC	<b>→</b>			<b>→</b>	BYE(REL)		
	REL	+			+	200 OK BYE(RLC)		
	REL_COM	<b>→</b>						

TP701004	SIP reference: RFC 3	261 [4]		SUP reference: Q.1912.5 [1],			
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP		Q.731	[i.2], clause 3.5.2.1.1			
SIP selection criteria	IODIV-(IOOI )-OII 700/OEII						
ISUP selection criteria							
Test purpose	Calling party number (user provided, verified and passed) with calling sub-address Verify that the IUT can successfully originate a call having a calling party number with the screening indicator set to "user provided, verified and passed" and an access transport parameter containing the calling sub-address.						
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body						
ISDN parameter values				·			
Comments	ISDN	S	UT	SIP-I			
	SETUP	<b>→</b>	→	INVITE(IAM)			
	ALERTING	+	+	180 Ringing(ACM)			
	CONN ← 200 OK INVITE(ANM)						
	→ ACK						
	Communication						
	DISC	<b>→</b>	→	BYE(REL)			
	REL	+	<b>+</b>	200 OK BYE(RLC)			
	REL_COM	<b>→</b>					

TP701005	SIP reference: RFC 3	261 [4]	0.7	ISUP reference: Q.1912.5 [1], 31 [i.2], clause 3.5.2.1.1			
TSS reference	ISDN-(ISUP)-SIP/SS/CLIP		Q.7.	51 [1.2], Clause 3.3.2.1.1			
SIP selection	10D14-(1001 )-011 /00/0211						
criteria							
ISUP selection criteria							
Test purpose	Calling party number (user provided, not verified)  Verify that the IUT can successfully originate a call having a default calling party number with the screening indicator set to "network provided" and a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified".						
SIP parameter values	INVITE: Content-Type: applica 180 Ringing: Content-Type: applica						
ISDN parameter values		•	•				
Comments	ISDN	5	SUT	SIP-I			
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)			
	ALERTING	+	<del>(</del>	180 Ringing(ACM)			
	CONN	+	<del>(</del>	200 OK INVITE(ANM)			
	→ ACK						
			unication				
	DISC	<b>→</b>	)	BYE(REL)			
	REL	+	+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>					

TP701006	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 3.5.2.1.1		
TSS reference	ISDN-(ISUP)-SIP/SS/0	CLIP					
SIP selection criteria							
ISUP selection criteria							
Test purpose	Calling party number (user provided, not verified) with calling sub-address Verify that the IUT can successfully originate a call having a default calling party number with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified" and an access transport parameter containing the calling sub-address.						
SIP parameter	INVITE: Content-Type:						
values	180 Ringing: Content-	Type: applicati	on/ISUP; /	ACM end	apsul	ated in the MIME body	
ISDN parameter values							
Comments	ISDN		SU	Γ		SIP-I	
	SETUP	→			<b>→</b>	INVITE(IAM)	
	ALERTING	+			+	180 Ringing(ACM)	
	CONN	+			+	200 OK INVITE(ANM)	
					<b>→</b>	ACK	
	Communication						
	DISC	<b>→</b>			<b>→</b>	BYE(REL)	
	REL	+			+	200 OK BYE(RLC)	
	REL_COM	<b>→</b>					

## A.1.1.2.2 Calling Line Identification Restriction (CLIR)

TP702001	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1		
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR						
SIP selection criteria							
ISUP selection criteria							
Test purpose	Restricted calling party number (network provided)  Verify that the IUT can successfully originate a call having a calling party number with the screening indicator set to "network provided" and the address presentation restricted indicator set to "presentation restricted".						
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body						
ISDN parameter values				-	•		
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONN ← 200 OK INVITE(ANM)						
	→ ACK						
		Communication					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	<b>→</b>		,			

TP702002	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1				
TSS reference	ISDN-(ISUP)-SIP/SS/CL	.IR						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Restricted calling party number (network provided) with calling sub-address Verify that the IUT can successfully originate a call having a calling party number with the screening indicator set to "network provided", the address presentation restricted indicator set to "presentation restricted" and an access transport parameter containing the calling sub-address.							
SIP parameter	INVITE: Content-Type: a	application/ISUP;	IAM encap	sulated i	n the MIME body			
values	180 Ringing: Content-Ty	pe: application/I	SUP; ACM	encapsul	ated in the MIME body			
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	<b>←</b>		+	200 OK INVITE(ANM)			
		→ ACK						
		Communication						
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	→						

TP702003	SIP reference: RFC	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1							
ISDN parameter values	ISDN-(ISUP)-SIP/SS/CLIR								
ISDN parameter values									
ISDN parameter values									
ISDN parameter values	Restricted calling party number (user provided, verified and passed)  Verify that the IUT can successfully originate a call having the calling party number with the screening indicator set to "user provided, verified and passed" and the address presentation restricted indicator set to "presentation restricted.								
ISDN parameter values	INVITE: Content-Type: application 180 Ringing: Content-Type:								
ISDN parameter values						·			
Comments	ISDN SETUP ALERTING CONN  DISC REL	→ ← ←	Communi	cation	→ ← ← → →	SIP-I INVITE(IAM) 180 Ringing(ACM) 200 OK INVITE(ANM) ACK BYE(REL) 200 OK BYE(RLC)			
	REL_COM	<b>→</b>							

TP702004	SIP reference: RFC 32	261 [4]	-	SUP reference: Q.1912.5 [1], [i.2], clause 4.5.2.1.1				
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Restricted calling party number (user provided, verified and passed) with calling sub-address  Verify that the IUT can successfully originate a call having a calling party number with the screening indicator set to "user provided, verified and passed", the address presentation restricted indicator set to "presentation restricted" and an access transport parameter containing the calling sub-address.							
SIP parameter	INVITE: Content-Type: applica	tion/ISUP; IAM	encapsulated i	n the MIME body				
values	180 Ringing: Content-Type: ap	plication/ISUP;	ACM encapsul	lated in the MIME body				
ISDN parameter values								
Comments	ISDN	SU	JT	SIP-I				
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)				
	ALERTING	+	<b>←</b>	180 Ringing(ACM)				
	CONN							
			<b>→</b>	ACK				
	Communication							
	DISC	<b>→</b>	<b>→</b>	BYE(REL)				
	REL	<del>-</del>	+	200 OK BYE(RLC)				
	REL_COM	<b>→</b>						

TP702005	SIP reference: RF0	C 3261 [4	]	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.5.2.1.1				
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Restricted calling party number (user provided, not verified)  Verify that the IUT can successfully originate a call having a default calling party number with the screening indicator set to "network provided" and a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted".							
SIP parameter	INVITE: Content-Type: app	lication/IS	SUP; IAM enca	psulated i	n the MIME body			
values	180 Ringing: Content-Type	: applicat	ion/ISUP; ACM	encapsu	lated in the MIME body			
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	Communication							
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

TP702006	SIP reference: RFC	3261 [4]		-	SUP reference: Q.1912.5 [1], [i.2], clause 4.5.2.1.1			
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR							
SIP selection criteria								
ISUP selection criteria								
Test purpose	Restricted calling party number (user provided, not verified) with calling sub-address  Verify that the IUT can successfully originate a call having a default calling party number with the screening indicator set to "network provided", a generic number containing the additional calling party number with the screening indicator set to "user provided, not verified", both having the address presentation restricted indicator set to "presentation restricted" and an access transport parameter containing the calling sub-address.							
SIP parameter	INVITE: Content-Type: applic							
values	180 Ringing: Content-Type: a	application	n/ISUP; ACM er	ncapsul	ated in the MIME body			
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	<b>←</b>		<b>←</b>	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	Communication							
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>	<u> </u>					

TP702007	SIP reference: RFC 32	261 [4]	-	SUP reference: Q.1912.5 [1],				
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR		Q.73	1 [i.2], clause 4.2.1				
SIP selection criteria	IODIN-(IOOI )-SII 703/GLIIX							
ISUP selection criteria	DLE							
Test purpose	Presentation of the address - interaction with MCID  Verify that the information conveyed in an incoming call (especially the calling party number and the additional calling party number in the generic number) is registered in the network regardless of whether the calling user has activated the CLIR service or not, if the called user has MCID activated.							
SIP parameter values	INVITE: Content-Type: applica 180 Ringing: Content-Type: ap							
ISDN parameter values			•	·				
Comments	ISDN	SU	JT	SIP-I				
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)				
	ALERTING	<b>←</b>	<b>←</b>	180 Ringing(ACM)				
	CONN	<del>-</del>	←	200 OK INVITE(ANM)				
			<b>→</b>	ACK				
		Communication						
	DISC	<b>→</b>	<b>→</b>	BYE(REL)				
	REL	+	+	200 OK BYE(RLC)				
	REL_COM	<b>→</b>						

TP702008	SIP reference: RFC 3261 [4]			I	SUP reference: Q.1912.5 [1],			
				Q.73	31 [i.2], clause 4.2.1			
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR							
SIP selection criteria								
ISUP selection criteria	DLE							
Test purpose	Presentation of the address - called party has override category  Verify that the calling party number and the additional calling party number in the  generic number are passed to the access regardless of whether the calling user has activated the CLIR service or not if the called user has the override category.							
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body							
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
	→ ACK							
		Communication						
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+	•	+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>	<del></del>					

TP702009	SIP reference: RFC	3261 [4	]	_	SUP reference: Q.1912.5 [1], 1 [i.2], clause 4.2.1				
TSS reference	ISDN-(ISUP)-SIP/SS/CLIR								
SIP selection criteria									
ISUP selection criteria	DLE								
Test purpose		Presentation of the address - called party has not override category Verify that the calling party number is not passed to the access.							
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body								
ISDN parameter values				•	•				
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Communication	•					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	<b>→</b>							

TP7020010	SIP reference: RFC 3261 [4]			(	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 4.2.1			
TSS reference	ISDN-(ISUP)-SIP/SS/CLII	R						
SIP selection								
criteria								
ISUP selection	DLE							
criteria								
Test purpose	Presentation of the address - called party has not override category Verify that the calling party number and the additional calling party number in the generic number are not passed to the access.							
SIP parameter	INVITE: Content-Type: ap	plication/IS	SUP; IAM e	ncapsula	ted ir	n the MIME body		
values	180 Ringing: Content-Typ	e: applicati	on/ISUP; A	CM enca	psul	ated in the MIME body		
ISDN parameter values								
Comments	ISDN		SUT			SIP-I		
	SETUP	→			<b>→</b>	INVITE(IAM)		
	ALERTING	<del>-</del>			<del>(</del>	180 Ringing(ACM)		
	CONN	+			<del>(</del>	200 OK INVITE(ANM)		
					<b>→</b>	ACK		
	Communication							
	DISC	<b>→</b>			<b>→</b>	BYE(REL)		
	REL	+			<del>(</del>	200 OK BYE(RLC)		
	REL_COM	<b>→</b>			•			

## A.1.1.2.3 Connected Line Identification Presentation (COLP)

TP703001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)			
TSS reference	ISDN-(ISUP)-SIP/SS/0	COLP	•				
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Initiate COLP reques	t	•	•			
	Verify that the exchange optional forward call		uccessfully a c	all reque	esting the COLP service in the		
SIP parameter	INVITE: Content-Type	: application/ISL	JP; IAM encap	sulated i	n the MIME body		
values					lated in the MIME body		
ISDN parameter values							
Comments	ISDN		SUT		SIP-I		
	SETUP	<b>→</b>		→	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
		C	Communication	)			
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE(RLC)		
	REL_COM	→					

TP703002	SIP reference: RFC	3261 [4	1]		SUP reference: Q.1912.5 [1], [i.2], clause 5.5.2.5.1 i)				
TSS reference	ISDN-(ISUP)-SIP/SS/COLP								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Verify that the IUT can provide	Connected number (user provided, verified and passed) Verify that the IUT can provide a connected number with the screening indicator set to "user provided, verified and passed", if the user provided COL is valid.							
SIP parameter	INVITE: Content-Type: appli								
values		180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body							
ISDN parameter values									
Comments	ISDN		SUT	Г	SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	CASE A								
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	<b>←</b>		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
	CASE B								
	CONN	+		<del>-</del>	200 OK INVITE(CON)				
				→	ACK				
			Communi						
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	→							

TP703003	SIP reference	: RFC 3261 [4]	I	ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)					
TSS reference	ISDN-(ISUP)-SIP/SS/0	COLP							
SIP selection criteria									
ISUP selection criteria									
Test purpose	address Verify that the IUT can "user provided, verified	Connected number (user provided, verified and passed) with connected sub- address  Verify that the IUT can provide a connected number with the screening indicator set to  "user provided, verified and passed", if the user provided COL is valid and an access  transport parameter containing the connected sub-address.							
SIP parameter values	INVITE: Content-Type 180 Ringing: Content-	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body							
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	CASE A								
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
	CASE B								
	CONN	+		+	200 OK INVITE(CON)				
				<b>→</b>	ACK				
			Communication	n					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	<b>→</b>	-						

TP703004	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)					
TSS reference	ISDN-(ISUP)-SIP/SS/COLP								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Verify that the IUT can provide	Connected number (network provided)  Verify that the IUT can provide a default connected number with the screening indicator set to "network provided", if the user provided COL is not valid.							
SIP parameter	INVITE: Content-Type: appli								
values	180 Ringing: Content-Type: 200 OK INVITE: Content-Type	180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body							
ISDN parameter values									
Comments	ISDN		SU.	Γ	SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	CASE A								
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	<b>←</b>		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
	CASE B								
	CONN	+		<b>←</b>	200 OK INVITE(CON)				
				<b>→</b>	ACK				
			Communi						
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	→							

TP703005	SIP reference: RFC	3261 [4	]	ISUP reference: Q.1912.5 [1],				
				Q.731	[i.2], clause 5.5.2.5.1 i)			
TSS reference	ISDN-(ISUP)-SIP/SS/COLP			,	-			
SIP selection criteria								
ISUP selection criteria								
Test purpose	Connected number (network provided) with connected sub-address Verify that the IUT can provide a default connected number with the screening indicator set to "network provided", if the user provided COL is not valid and an access transport parameter containing the connected sub-address.							
SIP parameter	INVITE: Content-Type: applic							
values		180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body						
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	→		<b>→</b>	INVITE(IAM)			
	CASE A							
	ALERTING	<b>←</b>		<b>←</b>	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
				<b>→</b>	ACK			
			Communication					
	DISC	→		<b>→</b>	BYE(REL)			
	REL	+		<del>-</del>	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

TP703006	SIP reference: RFC	3261 [4	l		SUP reference: Q.1912.5 [1], [i.2], clause 5.5.2.5.1 i)			
TSS reference	ISDN-(ISUP)-SIP/SS/COLP							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Connected number (user provided, not verified)  Verify that the IUT can provide a default connected number with the screening indicator set to "network provided" and a generic number containing the additional connected number with the screening indicator set to "user provided, not verified".							
SIP parameter	INVITE: Content-Type: applie							
values	180 Ringing: Content-Type: a 200 OK INVITE: Content-Type							
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	+	-	+	200 OK INVITE(CON)			
				<b>→</b>	ACK			
			Communica	ition				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

TP703007	SIP reference:	RFC 3261 [4]			ISUP reference: Q.1912.5 [1], [i.2], clause 5.5.2.5.1 i)				
TSS reference	ISDN-(ISUP)-SIP/SS/0	COLP							
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Verify that the IUT can set to "network provide with the screening indi	Connected number (user provided, not verified) with connected sub-address  Verify that the IUT can provide a default connected number with the screening indicator set to "network provided", a generic number containing the additional connected number with the screening indicator set to "user provided, not verified" and an access transport parameter containing the connected sub-address.							
SIP parameter values	INVITE: Content-Type 180 Ringing: Content-	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body							
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	CASE A								
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE(ANM)				
				→	ACK				
	CASE B								
	CONN	+		+	200 OK INVITE(CON)				
				→	ACK				
		Co	mmunication	1					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	<b>→</b>							

TP703008	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 5.5.2.5.1 i)				
TSS reference	ISDN-(ISUP)-SIP/SS/COLP							
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Verify that an exchange with	COLP - interaction with MSN Verify that an exchange with MSN can provide the connected party multiple subscriber						
SIP parameter values	INVITE: Content-Type: applic 180 Ringing: Content-Type: a	number or full ISDN number as the <b>connected number</b> on call answer.  INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body  180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body  200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body						
ISDN parameter values					,			
Comments	ISDN		SU	IT	SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CASE A							
	ALERTING	+		<del>(</del>	180 Ringing(ACM)			
	CONN	+		<b>←</b>	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	CASE B							
	CONN	<b>←</b>		+	200 OK INVITE(CON)			
	→ ACK							
			Commur	ication				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	<b>←</b>		<b>←</b>	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

### A.1.1.2.4 Connected Line Identification Restriction (COLR)

TP704001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 6.5.2.1.2				
TSS reference	ISDN-(ISUP)-SIP/SS/	COLR						
SIP selection								
criteria								
ISUP selection criteria								
Test purpose	Presentation of restricted COL  Verify that a local exchange will not pass the information on to the access signalling system when a <b>connected number</b> is received in the ANM or CON and its address presentation restricted indicator is set to "presentation restricted", i.e. that presentation is denied on the user-network interface (UNI).							
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body							
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
				<b>→</b>	ACK			
		С	ommunication	1				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

TP704002	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 6.5.2.1.2					
TSS reference	ISDN-(ISUP)-SIP/SS/0	COLR							
SIP selection criteria									
ISUP selection criteria									
Test purpose	Verify that the received number in the <b>generic</b> calling user, if the calle	Presentation of restricted COL to "override category" calling user  Verify that the received connected number and optionally the additional connected number in the generic number can be conveyed successfully to an "override category" calling user, if the called user has activated the Connected Line Presentation Restriction (COLR) supplementary service.							
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body								
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	→		<b>→</b>	INVITE(IAM)				
	CASE A								
	ALERTING	+		<b>←</b>	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
	CASE B								
	CONN	+		+	200 OK INVITE(CON)				
				<b>→</b>	ACK				
			Communication	1					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	<b>→</b>	<del></del>						

TP704003	SIP reference: RFC	3261 [4]			ISUP reference: Q.1912.5 [1], [i.2], clause 6.5.2.1.2				
TSS reference	ISDN-(ISUP)-SIP/SS/COLR								
SIP selection criteria									
ISUP selection criteria	DLE								
Test purpose	Verify that the IUT can provid "user provided, verified and p set to "presentation restricted	Restricted connected number (user provided, verified and passed)  Verify that the IUT can provide a connected number with the screening indicator set to  "user provided, verified and passed" and with the address presentation restricted indicator set to "presentation restricted", if the user provided COL is valid.							
SIP parameter	INVITE: Content-Type: applic								
values	180 Ringing: Content-Type: a 200 OK INVITE: Content-Typ								
ISDN parameter values									
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		→	INVITE(IAM)				
	CASE A								
	ALERTING	+		<b>←</b>	180 Ringing(ACM)				
	CONN	+		<b>←</b>	200 OK INVITE(ANM)				
				→	ACK				
	CASE B								
	CONN	<b>←</b>		<b>←</b>	200 OK INVITE(CON)				
				<b>→</b>	ACK				
		(	Communication	า					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	<b>→</b>							

TP704004	SIP reference:				ISUP reference: Q.1912.5 [1], 1 [i.2], clause 6.5.2.1.			
TSS reference	ISDN-(ISUP)-SIP/SS/C	OLR						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Restricted connected number (user provided, verified and passed) with connected sub-address  Verify that the IUT can provide a connected number with the screening indicator set to "user provided, verified and passed" and with the address presentation restricted indicator set to "presentation restricted", if the user provided COL is valid. Additionally, an access transport parameter containing the connected sub-address shall also be provided.							
SIP parameter values	INVITE: Content-Type: application/ISUP; IAM encapsulated in the MIME body 180 Ringing: Content-Type: application/ISUP; ACM encapsulated in the MIME body 200 OK INVITE: Content-Type: application/ISUP; ANM encapsulated in the MIME body							
ISDN parameter values					•			
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
	CASE B							
	CONN	+		+	200 OK INVITE(CON)			
	ACK							
		Comm	nunication	<u> </u>				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		<b>←</b>	200 OK BYE(RLC)			
	REL_COM	<b>→</b>						

TP704005	SIP reference: RFC	3261 [4	1		SUP reference: Q.1912.5 [1], 1 [i.2], clause 6.5.2.5				
TSS reference	ISDN-(ISUP)-SIP/SS/COLR								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Verify that the IUT can provided and set to "network provided" and	Restricted connected number (network provided)  Verify that the IUT can provide a default connected number with the screening indicator set to "network provided" and with the address presentation restricted indicator set to "presentation restricted", if the user provided COL is not valid.							
SIP parameter	INVITE: Content-Type: applie								
values	180 Ringing: Content-Type: a 200 OK INVITE: Content-Type								
ISDN parameter values									
Comments	ISDN		SUT	•	SIP-I				
	SETUP	<b>→</b>		→	INVITE(IAM)				
	CASE A								
	ALERTING	<b>←</b>		<b>←</b>	180 Ringing(ACM)				
	CONN	+		<b>←</b>	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
	CASE B								
	CONN	+		+	200 OK INVITE(CON)				
				→	ACK				
			Communio	cation					
	DISC	<b>→</b>		→	BYE(REL)				
	REL	+		<b>+</b>	200 OK BYE(RLC)				
	REL_COM	<b>→</b>							

TP704006	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 6.5.2.1.2					
TSS reference	ISDN-(ISUP)-SIP/SS/0	COLR							
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Verify that the IUT can set to "network provide number with the scree	Restricted connected number (user provided, not verified) Verify that the IUT can provide a default connected number with the screening indicator set to "network provided" and a generic number containing the additional connected number with the screening indicator set to "user provided, not verified" - both having the address presentation restricted indicator set to "presentation restricted".							
SIP parameter	INVITE: Content-Type								
values	180 Ringing: Content-	Type: application	/ISUP; ACM	encapsu	psulated in the MIME body				
ISDN parameter values		71 11	,		,				
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	CASE A								
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
	CASE B								
	CONN	+		+	200 OK INVITE(CON)				
				→	ACK				
		C	ommunicatio	n .					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE(RLC)				
	REL_COM	<b>→</b>							

TP704007	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], Q.731 [i.2], clause 6.5.2.1.2				
TSS reference	ISDN-(ISUP)-SIP/SS/CC	)LR						
SIP selection criteria								
ISUP selection criteria								
Test purpose	Restricted connected number (user provided, not verified) with connected subaddress  Verify that the IUT can provide a default calling party number with the screening indicator set to "network provided", a generic number containing the additional connected number with the screening indicator set to "user provided, not verified" - both having the address presentation restricted indicator set to "presentation restricted" and additionally an access transport parameter containing the connected sub-address.							
SIP parameter values	INVITE: Content-Type: a 180 Ringing: Content-Ty	application/ISUpe: application	JP; IAM enca n/ISUP; ACN	apsulated I encapsu	in the MIME body			
ISDN parameter values								
Comments	ISDN		SUT		SIP-I			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	CASE A							
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	<del>-</del>		+	200 OK INVITE(ANM)			
				→	ACK			
	CASE B							
	CONN	<del>-</del>		+	200 OK INVITE(CON)			
				→	ACK			
			Communicati					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	<b>←</b>		+	200 OK BYE(RLC)			
	REL_COM	→						

## A.1.1.2.5 Terminal Portability (TP)

TP705001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], 4.5.2.1.1 a)/Q.733 [i.6]				
TSS reference	ISDN-(ISUP)-SIP/SS/TF	•						
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that the calling	Terminal portability, requested by the calling party To verify that the calling party can suspend and resume an outgoing call and that user initiated SUS and RES messages are sent to the succeeding exchange.						
SIP parameter					<u> </u>			
values								
ISDN parameter								
values								
Comments	ISDN		SU <sup>.</sup>	Т	SIP			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
		Co	mmuni	ication				
	SUSPEND	<b>→</b>		<b>→</b>	INFO(SUS)			
				+	200 OK INFO			
	RESUME	<b>→</b>		<b>→</b>	INFO(RES)			
				+	200 OK INFO			
		Co	mmuni	ication				
	DISC	<b>→</b>		→	BYE(REL)			
	REL	+		+	200 OK BYE			
	REL_COM	<b>→</b>	•					

TP705002	SIP reference	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], 4.5.2.1.1 b)/Q.733 [i.6]			
TSS reference	ISDN-(ISUP)-SIP/SS/	TP						
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that IUT info	Terminal portability, requested by the called party To verify that IUT informs the calling party that a suspend and a resume have been requested by the called party upon receipt of user initiated SUS and RES messages.						
SIP parameter								
values								
ISDN parameter								
values								
Comments	ISDN		SUT		SIP			
	SETUP	→		→	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
		Co	ommunicatio	n				
	SUSPEND	+		+	INFO(SUS)			
				<b>→</b>	200 OK INFO			
	RESUME	+		+	INFO(RES)			
				<b>→</b>	200 OK INFO			
		Co	ommunicatio	n				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE			
	REL_COM	<b>→</b>						

TP705003	SIP reference: RI	FC 3261 [4]			ISUP reference: Q.1912.5 [1], 4.5.2.1.2/Q.733 [i.6]
TSS reference	ISDN-(ISUP)-SIP/SS/TP				
SIP selection criteria					
ISUP selection criteria					
Test purpose		eleased with	cause #1	02 (recovery	o Resume after Suspend of on timer expiry) by the IUT if sume the call.
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SU	Γ	SIP
	SETUP	<b>→</b>		→	INVITE(IAM)
	ALERTING	+		+	180 Ringing(ACM)
	CONN	+		<b>←</b>	200 OK INVITE(ANM)
		C	Communi	cation	
	SUSPEND	<b>→</b>		→	INFO(SUS)
				+	200 OK INFO
			T2 exp	oiry	
				<b>→</b>	BYE(REL)
				+	200 OK BYE

TP705004	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], 4.5.2.1.1/Q.733 [i.6]			
TSS reference	ISDN-(ISUP)-SIP/SS/TP						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	Terminal portability, rel				·		
		ed call can b	e released by the	e IUT, if	the local user or the remote		
	user releases the call.						
SIP parameter							
values							
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ALERTING	<b>←</b>		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
	Communication						
	SUSPEND	→		<b>→</b>	INFO(SUS)		
				+	200 OK INFO		
				+	BYE(REL)		
			_	<b>→</b>	200 OK BYE		

TP705005	SIP reference: R	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], 4.5.2.5.1 a)/Q.733 [i.6]				
TSS reference	ISDN-(ISUP)-SIP/SS/TP								
SIP selection criteria									
ISUP selection criteria									
Test purpose	To verify that the IUT info	Terminal portability, requested by the calling party To verify that the IUT informs the called party that suspend and resume have been requested by the calling party upon receipt of user initiated SUS and RES messages.							
SIP parameter									
values									
ISDN parameter									
values					1				
Comments	ISDN		SUT		SIP				
	SETUP	+		+	INVITE(IAM)				
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)				
	CONN	→		→	200 OK INVITE(ANM)				
		C	ommunicatio	n					
	NOTIFY(suspend)	+		+	INFO(SUS)				
				<b>→</b>	200 OK INFO				
	NOTIFY(resume)	+		+	INFO(RES)				
				<b>→</b>	200 OK INFO				
		Co	ommunicatio	n					
	DISC	+		+	BYE(REL)				
	REL	<b>→</b>		<b>→</b>	200 OK BYE				
	REL_COM	+							

TP705006	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], 4.5.2.5.1 b)/Q.733 [i.6]		
TSS reference	ISDN-(ISUP)-SIP/SS/TP					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Terminal portability, red To verify that the called p initiated SUS and RES m	oarty can susp	end and resur	ne an in	coming call and that user exchange.	
SIP parameter						
values						
ISDN parameter values						
Comments	ISDN		SUT		SIP	
	SETUP	+		+	INVITE(IAM)	
	ALERTING	→		<b>→</b>	180 Ringing(ACM)	
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)	
		Co	ommunicatio	า		
	NOTIFY(suspend)	<b>→</b>		<b>→</b>	INFO(SUS)	
				+	200 OK INFO	
	NOTIFY(resume)	<b>→</b>		<b>→</b>	INFO(RES)	
				+	200 OK INFO	
		Co	ommunicatio	า		
	DISC	+		+	BYE(REL)	
	REL	<b>→</b>		<b>→</b>	200 OK BYE	
	REL_COM	+	•			

## A.1.1.2.6 User-to-User Signalling (UUS)

### A.1.1.2.6.1 User-to-User Signalling Service 1 (UUS1)

TP706001	SIP reference: RF	C 3261 [4]	]			ISUP reference:
						Q.1912.5 [1],
						1.1.5.2.1.1.1; 1.1.5.2.1.1.3;
					<u>1.1.</u>	5.2.2-4.1/Q.737 [i.10]
TSS reference	ISDN-(ISUP)-SIP/SS/UUS	1				
SIP selection criteria						
ISUP selection						
criteria						
Test purpose	UUS1 implicit - request					
						vith an UUS 1 implicit request,
	having the user-to-user in	nformation	<b>n</b> paramet	er in the <b>L</b>	AM,	without the <b>user-to-user</b>
	indicators parameter.					
SIP parameter						
values						
ISDN parameter values	SETUP: User-to-user infor	mation				
Comments	ISDN		SUT			SIP
	SETUP(UUInf)	<b>→</b>		7	<b>&gt;</b>	INVITE(IAM UUInf)
	ALERTING	+		+		180 Ringing(ACM)
	CONN(UUInf)	+		+	•	200 OK INVITE(ANM)
				7	<b>)</b>	ACK
		0	Communic	cation		
	DISC	+			<del>(</del>	BYE(REL)
	REL	<b>→</b>			<b>→</b>	200 OK BYE
	REL_COM	+				

TP706002	SIP reference: RFC 3261 [4]			ISUP reference:
				Q.1912.5 [1],
			clauses	s 1.1.5.2.5.2.3; 1.1.5.2.2-4.2/
				Q.737 [i.10]
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1			
SIP selection				
criteria				
ISUP selection				
criteria				
Test purpose	UUS1 implicit - discarded with indi			
	To verify that the IUT can, after succe			
	request, continue normal call set up i			
	user-to-user indicators set to "user	-to-user in	formation di	iscarded by the network" (see
	note).			
SIP parameter				
values				
ISDN parameter				
values				1
Comments	ISDN	SUT		SIP
	SETUP(UUInf) →		<b>→</b>	INVITE(IAM UUInf)
	ALERTING ←		<b>←</b>	180 Ringing(ACM UUInd)
	CONN +		<b>←</b>	200 OK INVITE(ANM)
			→	ACK
		communic	ation	
	DISC ←		+	BYE(REL)
	REL →		<b>→</b>	200 OK BYE
	REL_COM ←			
NOTE: The user-	to-user information is discarded becau	use the foll	owing netw	ork does not support it.

TP706003	SIP reference:	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], 1.1.5.2.5.2.3; 1.1.5.2.3-5.2/ Q.737 [i.10]			
TSS reference	ISDN-(ISUP)-SIP/SS/L	JUS1						
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that the IUT of	UUS1 implicit - discarded but no indication received  To verify that the IUT can successfully initiate/transit a call with an UUS1 implicit request, and complete the call if no indication is provided in the backward direction (see note).						
SIP parameter								
values								
ISDN parameter								
values								
Comments	ISDN		SUT		SIP			
	SETUP(UUInf)	→		<b>→</b>	INVITE(IAM UUInf)			
	ALERTING	+		+	180 Ringing(ACM)			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
		Co	ommunicati	ion				
	DISC	+		<b>+</b>	BYE(REL)			
	REL	<b>→</b>		<b>→</b>	200 OK BYE			
	REL_COM	+						
1) the	er-to-user information is di remote network is unable remote user may not be a	to pass the ser	vice 1 in any					

TP706004	SIP reference:	RFC 3261	[4]	cla	ISUP reference: Q.1912.5 [1], nuses 1.1.5.2.1.1.1; 1.1.5.2.1.1.3; 1.1.5.2.3-5.1/Q.737 [i.10]
TSS reference	ISDN-(ISUP)-SIP/SS/U	US1			
SIP selection					
criteria					
ISUP selection criteria					
Test purpose		an success user-to-u	ser <sup>°</sup> informa	<b>ation</b> pa	a call with an UUS1 implicit request, arameter in the ACM, CPG, ANM or cors).
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP(UUInf)	<b>→</b>		<b>→</b>	INVITE(IAM UUInf)
	CASE A				
	ALERTING(UUInf)	+		+	180 Ringing(ACM UUInf)
	CASE B				
	ALERTING	+		+	180 Ringing(ACM)
	NOTIFY(UUInf)			+	183 Session Progress(CPG UUInf)
	CASE C				
	CONN(UUInf)	+		+	200 OK INVITE(CON UUInf)
				<b>→</b>	ACK
	CASE D				
	ALERTING	+		+	180 Ringing(ACM)
	CONN(UUInf)	+		+	200 OK INVITE(ANM UUInf)
				<b>→</b>	ACK
		Co	mmunicat	ion	
	DISC	+		+	BYE(REL)
	REL	<b>→</b>		<b>→</b>	200 OK BYE
	REL_COM	+			

TP706005	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 1.1.5.2.5.2.3; 1.1.5.2.3-5.2/ Q.737 [i.10]					
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1					-			
SIP selection criteria									
ISUP selection criteria									
Test purpose	UUS1 implicit - discard wit To verify that the IUT can su and set the user-to-user ind network" in the first backward	ccessfu dicators	Ily transit/a to "user-t	accept a o-user i	nforma				
SIP parameter values			<b>3</b> - 7						
ISDN parameter values									
Comments	SIP		SU	Т		ISDN			
	INVITE(IAM UUInf)	<b>→</b>			→	SETUP			
	180 Ringing(ACM UUInd)	+			+	ALERTING			
	200 OK INVITE(ANM)	+			+	CONN			
	ACK	<b>→</b>							
		Communication							
	BYE(REL)	+			+	DISC			
	200 OK BYE	<b>→</b>			<b>→</b>	REL			
					+	REL_COM			
NOTE: The user-	to-user information is discarde	ed beca	use the ne	etwork d	oes no	t support it.			

TP706006	SIP reference: RFC 32	261 [4]		claus	-	SUP reference: Q.1912.5 [1], 1.5.2.1.1.2; 1.1.5.2.2-4.1/ Q.737 [i.10]
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1					
SIP selection criteria						
ISUP selection criteria						
Test purpose	To verify that the IUT can succ non-essential request, by inclu and the user-to-user indicato	essfully ding/tra	/ initiate/tr insferring	the <b>use</b>	r-to-u	ser information parameter
SIP parameter values						
ISDN parameter values						
Comments	ISDN		SU'	Γ		SIP
	SETUP(FAC uus1reqness)					INVITE(IAM UUInd, UUInf)
	ALERTING(FAC uus1rr)					180 Ringing(ACM)
	CONN					200 OK INVITE(ANM)
						ACK
		(	Communi	cation		
	BYE(REL)	+			+	DISC
	200 OK BYE	<b>→</b>			<b>→</b>	REL
			, and the second		+	REL_COM

TP706007	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 1.1.5.2.5.2.3, 1.1.5.2.2-4.2/ Q.737 [i.10]				
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1								
SIP selection									
criteria									
ISUP selection									
criteria					-				
Test purpose	non-essential request, and co	ccessfu ontinue dicato	Illy initiate normal c <b>rs</b> param	e/tran all se eter i	received sit a call with an UUS1 explicit et up if the UUS1 service is explicitly is received as "service not provided" in				
SIP parameter values									
ISDN parameter values									
Comments	ISDN	ļ	SUT	<u> </u>	SIP				
	SETUP(FAC uus1reqness)	<b>→</b>	<u> </u>	<b>→</b>	INVITE(IAM UUInd, UUInf)				
	CASE A	<u> </u>		<u> </u>					
	ALERTING(FAC uus1rr)	+		+	180 Ringing(ACM UUInd s1 prov)				
	CONN	+	<u> </u>	+	200 OK INVITE				
	CASE B	ļ	ļ	<u> </u>					
		<u> </u>	ļ	+	183 Session Progress(ACM)				
	ALERTING(FAC uus1rr)	+		<b>←</b>	180 Ringing(CPG UUInd s1 prov)				
	CONN	+	ļ	+	200 OK INVITE				
	CASE C	ļ		ļ					
	ALERTING	+		+	180 Ringing(ACM)				
	CONN(FAC uus1rr)	+		<b>←</b>	200 OK INVITE(ANM UUInd s1 prov)				
	CASE D	ļ	<u> </u>	ļ					
	CONN(FAC uus1rr)	+	<u> </u>	<b>←</b>	200 OK INVITE(ANM UUInd s1 prov)				
			municati						
	BYE(REL)	<b>←</b>		+	DISC				
	200 OK BYE	<b>→</b>	REL						
i				<b>←</b>	REL_COM				

- the network is unable to pass the explicit service 1 in any message;
   the remote user may not be able to interpret incoming UUS information.

TP706008	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses1.1.5.2.5.2.3, 1.1.5.2.2-4.2/ Q.737 [i.10]			
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1		•				
SIP selection criteria							
ISUP selection criteria							
Test purpose	UUS1 explicit non-essential - implicit (no explicit) rejection received  To verify that the IUT can successfully initiate/transit a call with an UUS1 explicit non-essential request, and continue normal call set up if no indication is provided in the backward direction (see note).						
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP(FAC uus1reqness)	<b>→</b>		<b>→</b>	INVITE(IAM UUInd, UUInf)		
	ALERTING(FAC uus1rr)	+		+	180 Ringing(ACM)		
	CONN(FAC uus1reterr)	+		+	200 OK INVITE(ANM)		
				<b>→</b>	ACK		
		C	ommunicati	on			
	BYE(REL)	<del>(</del>		+	DISC		
	200 OK BYE → REL						
	<b>←</b> REL_COM						
1) the ne	-to-user information is discarded because: etwork is unable to pass the explicit service 1 in any message; emote user may not be able to interpret incoming UUS information.						

TP706009	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 1.1.5.2.1.1.2, 1.1.5.2.3-5.1/ Q.737 [i.10]			
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1						
SIP selection							
criteria							
ISUP selection							
criteria							
Test purpose	UUS1 explicit non-essential - acce To verify that the IUT can successful non-essential request, by transferring the ACM, CPG, ANM or CON set to	ly transi g/includ	it/accept a ing the <b>use</b>	r-to-u			
SIP parameter values			-				
ISDN parameter							
values							
Comments	SIP		SUT		ISDN		
	INVITE(IAM UUInd, UUInf)	→		→	SETUP(FAC uus1reqness)		
					CASE A		
	180 Ringing(ACM UUInd s1 prov)	<b>←</b>		+	ALERTING(FAC uus1rr)		
	200 OK INVITE	+		+	CONN		
					CASE B		
	183 Session Progress(ACM)	+					
	180 Ringing(CPG UUInd s1 prov)	+		+	ALERTING(FAC uus1rr)		
	200 OK INVITE	<b>←</b>		+	CONN		
					CASE C		
	180 Ringing(ACM)	+		+	ALERTING		
	200 OK INVITE(ANM UUInd s1	+		+	CONN(FAC uus1rr)		
	prov)						
					CASE D		
	200 OK INVITE(ANM UUInd s1 ← CONN(FAC uus1rr)						
	,	Co	mmunicati	ion			
	DISC	+		+	BYE(REL)		
	REL	<b>→</b>		<b>→</b>	200 OK BYE		
	REL_COM			+			

TP706010	SIP reference: RFC 3	3261 [4	-		SUP reference: Q.1912.5 [1], .1.5.2.5.2.2, 1.1.5.2.2-5.2/ Q.737 [i.10]
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1				
SIP selection criteria					
ISUP selection criteria					
Test purpose	UUS1 explicit non-essentia To verify that the IUT can transequest, and reject the service the ACM, CPG, ANM or CON	nsfer/a e by no	ccept a call with an	ŪUS	
SIP parameter values			,		
ISDN parameter values					
Comments	SIP		SUT		ISDN
	INVITE(IAM UUInd, UUInf)	<b>→</b>		<b>→</b>	SETUP(FAC uus1reqness)
	180 Ringing(ACM)	+		+	ALERTING
	200 OK INVITE(ANM)	+		+	CONN
	ACK	<b>→</b>			
			Communication		
	DISC	+		+	BYE(REL)
	REL	<b>→</b>		<b>→</b>	200 OK BYE
	REL_COM			+	
NOTE: The netwo	ork or the user cannot support	UUS1.			

TP706011	SIP reference: RFC 32	261 [4]	clau	-	SUP reference: Q.1912.5 [1], 1.5.2.1.1.2, 1.1.5.2.2-5.1/ Q.737 [i.10]			
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1							
SIP selection criteria								
ISUP selection criteria								
Test purpose	UUS1 explicit essential - request To verify that the IUT can successfully originate/transit a call having an UUS1 explicit essential request, by including/transferring in the IAM the user-to-user information parameter, the user-to-user indicators set to "request, essential" and the ISDN user part preference indicator in the forward call indicators set to "ISUP required all the way".							
SIP parameter					. ,			
values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP(FAC uus1reqess)				INVITE(IAM UUInd, UUInf)			
	ALERTING(FAC uus1rr)				180 Ringing(ACM)			
	CONN				200 OK INVITE(ANM)			
					ACK			
			Communication					
	BYE(REL)	+		+	DISC			
	200 OK BYE	<b>→</b>		<b>→</b>	REL			
				+	REL_COM			

TP706012	SIP reference: RFC	3261 [4	]	clauses	ISUP reference: Q.1912.5 [1], 1.1.5.2.5.2.2, 1.1.5.2.2-5.2/ Q.737 [i.10]			
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1							
SIP selection criteria								
ISUP selection criteria								
Test purpose	UUS1 explicit essential - implicit rejection (no explicit acceptance received)  To verify that the service can be rejected if no indication (no user-to-user indicators parameter or the service 1 field in the user-to-user indicators set to "no information" or "not provided") is received in the first backward message (implicit rejection of service 1) (see note).							
SIP parameter values								
ISDN parameter values								
Comments	ISDN		SUT	•	SIP			
	SETUP(FAC uus1reqess)	<b>→</b>		<b>→</b>	INVITE(IAM UUInd, UUInf)			
	ALERTING(FAC uus1rr)	+		+	180 Ringing(ACM)			
	CONN(FAC uus1reterr)	+		+	200 OK INVITE(ANM)			
				→	ACK			
			Communic	cation				
	BYE(REL)	+		+	DISC			
	200 OK BYE	<b>→</b>		<b>→</b>	REL			
				+	REL_COM			
NOTE: The netw	ork does not understand the s	ervice 1	request. In	this case th	e call should be released.			

TP706013	SIP reference: RFC 3261 [4	claus	ISUP reference: Q.1912.5 [1], clauses 1.1.5.2.1.1.2, 1.1.5.2.2-5.1/ Q.737 [i.10]					
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose	UUS1 explicit essential - acceptance To verify that the IUT can successfully complete a call with an UUS1 explicit essential request having the user-to-user indicators parameter in the ACM, CPG, ANM or CON set to "service provided".							
SIP parameter values								
ISDN parameter								
values								
Comments	SIP		SUT		ISDN			
	INVITE(IAM UUInd, UUInf)	→		<b>→</b>	SETUP(FAC uus1reqess)			
	CASE A							
	180 Ringing(ACM UUInd s1 prov)	<b>←</b>		+	ALERTING(FAC uus1rr)			
	200 OK INVITE	<b>←</b>		+	CONN			
	CASE B							
	183 Session Progress(ACM)	←						
	180 Ringing(CPG UUInd s1 prov)	<b>←</b>		+	ALERTING(FAC uus1rr)			
	200 OK INVITE	<b>←</b>		<b>←</b>	CONN			
	CASE C							
	180 Ringing(ACM)	<b>←</b>		+	ALERTING			
	200 OK INVITE(ANM UUInd s1 prov)	+		<b>←</b>	CONN(FAC uus1rr)			
	CASE D							
	200 OK INVITE(ANM UUInd s1	+	CONN(FAC uus1rr)					
	prov)							
	7100		mmunicatio					
	DISC	<b>+</b>		+	BYE(REL)			
	REL	<b>→</b>		<b>→</b>	200 OK BYE			
	REL_COM			<b>←</b>				

TP706014	SIP reference: RFC 3	3261 [4]		cla	ISUP reference: Q.1912.5 [1], auses 1.1.5.2.5.2.2, 1.1.5.2.2-5.2/ Q.737 [i.10]			
TSS reference	ISDN-(ISUP)-SIP/SS/UUS1							
SIP selection criteria								
ISUP selection criteria								
Test purpose	UUS1 explicit essential - rejection  To verify that the service can be rejected with a REL having the Cause value 29 "facility rejected" or 69 "requested facility not implemented", either with diagnostics (specifying the name of the user-to-user indicator parameter) (see note).							
SIP parameter values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP(FAC uus1reqess)	<b>→</b>		<b>→</b>	INVITE(IAM UUInd, UUInf)			
	RELEASE	+		+	500 Server Internal Error(REL#29)			
	RELEASE COLMPLETE	<b>→</b>		<b>→</b>	ACK			
NOTE: The netw	ork or the called user cannot so	upport tl	he service.					

## A.1.1.2.6.2 User-to-User Signalling Service 2 (UUS2)

TP706101	SIP reference:	RFC 3261 [4]	cla		ISUP reference: Q.1912.5 [1], .2.5.2.1.1.2, 1.2.5.2.2-5.1/ Q.737 [i.10]					
TSS reference	ISDN-(ISUP)-SIP/SS/U	US2								
SIP selection criteria										
ISUP selection criteria										
Test purpose	To verify that the IUT control essential request, having essential". To verify that	UUS2 explicit non-essential - request and acceptance To verify that the IUT can successfully originate/transit a call with an UUS2 explicit non-essential request, having the user-to-user indicators in the IAM set to "request, not essential". To verify that the IUT can successfully complete a call with an UUS2 explicit non-essential request, having the user-to-user indicators parameter in the ACM or CPG								
SIP parameter values										
ISDN parameter values										
Comments	ISDN		SUT		SIP					
	SETUP	→		<b>→</b>	INVITE(IAM UU2 not ess)					
	ALERTING	+		+	180 Ringing(ACM)					
	USER INFO	<b>→</b>		<b>→</b>	INFO(USR)					
				+	200 OK INFO					
	USER INFO	+		+	INFO(USR)					
				<b>→</b>	200 OK INFO					
	CONN	+		+	200 OK INVITE(ANM)					
		С	ommunicatior	1						
	DISC	+		+	BYE(REL)					
	RELEASE	<b>→</b>	<u> </u>	<b>→</b>	200 OK BYE					
	REL_COM	+								

TP706102	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clauses 1.2.5.2.5.2.2, 1.2.5.2.2-5.2/ Q.737 [i.10]					
TSS reference	ISDN-(ISUP)-SIP/SS/UUS2									
SIP selection criteria										
ISUP selection criteria										
Test purpose	To verify that the UUS2 service	UUS2 explicit non-essential - explicit rejection (service not provided) To verify that the UUS2 service can be rejected and the user-to-user indicators in the ACM or CPG are set to "service 2 not provided" (see note).								
SIP parameter			-		·					
values										
ISDN parameter										
values										
Comments	ISDN		SUT		SIP					
	SETUP(FAC uus2reqness)	<b>→</b>		<b>→</b>	INVITE(IAM UU2 ness)					
	CASE A									
	ALERTING(FAC uus1rr)	+		+	180 Ringing(ACM UUInd s2 prov)					
	CONN	+		+	200 OK INVITE					
	CASE B									
				+	183 Session Progress(ACM)					
	ALERTING(FAC uus1rr)	+		+	180 Ringing(CPG UUInd s1 prov)					
	CONN	+		+	200 OK INVITE					
		Com	municat	ion						
	BYE(REL)	+		<b>←</b>	DISC					
	200 OK BYE(RLC)	<b>→</b>		<b>→</b>	REL					
	,			+	REL_COM					
NOTE: The netw	ork or the user cannot support	UUS2.	•		•					

TP706103	SIP reference: RFC 3	3261 [4	•	ISUP reference: Q.1912.5 [1], clauses 1.2.5.2.5.2.3, 1.2.5.2.2-5.2/ Q.737 [i.10]					
TSS reference	ISDN-(ISUP)-SIP/SS/UUS2								
SIP selection criteria									
ISUP selection criteria									
Test purpose	UUS2 explicit non-essential - implicit rejection (no indication)  To verify that the IUT can successfully complete a call with an UUS2 explicit non-essential request, if no indication is provided in the backward direction (see note).								
SIP parameter values					,				
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP(FAC uus1reqness)	<b>→</b>		<b>→</b>	INVITE(IAM UU2 ness)				
	ALERTING(FAC uus1rr)	+		+	180 Ringing(ACM)				
	CONN(FAC uus1reterr)	+		+	200 OK INVITE(ANM)				
				1	ACK				
		Communication							
	BYE(REL)	+		+	DISC				
	200 OK BYE	<b>→</b>		<b>→</b>	REL				
				+	REL_COM				
NOTE: The netwo	ork or the user cannot support	UUS2.							

TP706104	SIP reference	e: RFC 3261 [4]	c	ISUP reference: Q.1912.5 [1], clauses 1.2.5.2.1.1.2, 1.2.5.2.2-5.1/ Q.737 [i.10]					
TSS reference	ISDN-(ISUP)-SIP/SS/	UUS2							
SIP selection criteria									
ISUP selection criteria									
Test purpose	To verify that the IUT essential request, have	UUS2 explicit essential - request To verify that the IUT can successfully originate/transit a call having an UUS2 explicit essential request, having the user-to-user indicators set to "request, essential" and the ISDN user part preference indicator of the forward call indicators in the IAM set to "ISUP required"							
SIP parameter									
values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM UU2 ess)				
	ALERTING	+		+	180 Ringing(ACM)				
	USER INFO	<b>→</b>		<b>→</b>	INFO(USR)				
				+	200 OK INFO				
	USER INFO	+		+	INFO(USR)				
				<b>→</b>	200 OK INFO				
	CONN	+		+	200 OK INVITE(ANM)				
		C	ommunication	n					
	DISC	+		+	BYE(REL)				
	RELEASE	<b>→</b>		<b>→</b>	200 OK BYE				
	REL_COM	<del>(</del>							

TP706105	SIP reference	: RFC 3261 [4]	I	clau	ISUP reference: Q.1912.5 [1], ses 1.2.5.2.1.1.2, 1.2.5.2.2-5.1/ Q.737 [i.10]		
TSS reference	ISDN-(ISUP)-SIP/SS/	UUS2					
SIP selection criteria							
ISUP selection criteria							
Test purpose	UUS2 explicit essential - acceptance To verify that the IUT can successfully complete a call having an UUS2 explicit essential request having the user-to-user indicators parameter in the ACM or CPG set to "service provided".						
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP	+		+	INVITE(IAM UU2 ess)		
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)		
	USER INFO	+		+	INFO(USR)		
				<b>→</b>	200 OK INFO		
	USER INFO	<b>→</b>		<b>→</b>	183 Session Progress(USR)		
	CONN	→		<b>→</b>	200 OK INVITE(ANM)		
		Co	mmunicati	on			
	DISC	+		<b>+</b>	BYE(REL)		
	RELEASE	→		<b>→</b>	200 OK BYE		
	REL_COM	+					

TP706106	SIP referen	nce: RI	RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clauses 1.2.5.2.5.2.1, 1.2.5.2.2-5.2/ Q.737 [i.10]			
TSS reference	ISDN-(ISUP)-SIP/S	SS/UUS	32					
SIP selection criteria								
ISUP selection criteria								
Test purpose	UUS2 explicit essential - rejection  To verify that the service can be rejected with a REL with the Cause value 29 "facility rejected" or 69 "requested facility not implemented" or value 88 "incompatible destination", all with diagnostics (user-to-user indicators name).							
SIP parameter values								
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	+		+	INVITE(IAM UU2 ess)			
	RELEASE	<b>^</b>		<b>→</b>	500 Server Internal error(REL#29, 69, 88)			
	REL_COM	+		+	ACK			

TP706107			FC 3261 [4	ij	ISUP reference: Q.1912.5 [1], clauses 1.2 5.2.5.2.1, 1.2.5.2.2-5.2/ Q.737 [i.10]			
TSS reference	ISDN-(ISUP)-SIF	P/SS/UUS	S2					
SIP selection criteria								
ISUP selection criteria								
Test purpose	UUS2 explicit essential - implicit rejection To verify that the service can be rejected if no indication is received (no user-to-user indicators parameter) in the first backward message (implicit rejection of service 2) (see note).							
SIP parameter values	180 Ringing: the	encapsu	lated ACM	l does	not contain an user-to-user response indicator			
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM UU2 ess)			
				+	180 Ringing(ACM)			
	RELEASE	+		<b>→</b>	CANCEL(REL)			
	REL_COMP	<b>→</b>		+	200 OK CANCEL			
				+	487 Request Terminated			
				<b>→</b>	ACK			
NOTE: The remo	te network does n	ot under	stand the s	service	2 request or the remote user cannot support			

## A.1.1.2.6.3 User-to-User Signalling Service 3 (UUS3)

TP706201	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], auses 1.3.5.2.1.1.2, 1.3.5.2.2-5.1/ Q.737 [i.10]				
TSS reference	ISDN-(ISUP)-SIP/SS/UUS3								
SIP selection criteria									
ISUP selection criteria									
Test purpose	UUS3 explicit non-essential - request and acceptance To verify that the IUT can successfully originate/transit a call with an UUS3 explicit non-essential request, having the user-to-user indicators in the IAM set to "request, not essential".  To verify that the IUT can successfully complete a call with an UUS3 explicit non-essential request, having the Service 3 field in the user-to-user indicators parameter in the ANM or CON set to "service provided".								
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP(UU3 req not ess)	<b>→</b>		<b>→</b>	INVITE(IAM UU3 not ess)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONN(UU3 ret res)	+		+	200 OK INVITE(ANM UU3 prov)				
		Co	mmunicat	ion					
	USER INFO	<b>→</b>		→	INFO(USR)				
				+	200 OK INFO				
	USER INFO	+		+	INFO(USR)				
				→	200 OK INFO				
	DISC	<b>→</b>		→	BYE(REL)				
	RELEASE	+		+	200 OK BYE				
	REL_COM	<b>→</b>							

TP706202	SIP reference: RFC 3261 [4]			cla	ISUP reference: Q.1912.5 [1], auses 1.3.5.2.1.1.2, 1.3.5.2.2-5.1/ Q.737 [i.10]		
TSS reference	ISDN-(ISUP)-SIP/SS/UUS	33					
SIP selection							
criteria							
ISUP selection							
criteria Test purpose							
	UUS3 explicit essential - request and acceptance To verify that the IUT can successfully originate/transit a call with an UUS3 explicit essential request, having in the IAM the user-to-user indicators set to "request, essential" and the ISDN user part preference indicator in the forward call indicators set to "ISUP required all the way".  To verify that the IUT can successfully complete a call with an UUS3 explicit essential request having in the ANM or CON the Service 3 field of the user-to-user indicators parameter set to "service provided".						
SIP parameter			-				
values							
ISDN parameter							
values							
Comments	ISDN		SUT		SIP		
	SETUP(UU3 req ess)	→		→	INVITE(IAM UU3 ess)		
	ALERTING	+		<b>←</b>	180 Ringing(ACM)		
	CONN(UU3 ret res)	+		+	200 OK INVITE(ANM UU3 prov)		
			ommunicat	ation			
	USER INFO	<b>→</b>		→	INFO(USR)		
				<b>←</b>	200 OK INFO		
	USER INFO	+		<b>←</b>	INFO(USR)		
				<b>→</b>	200 OK INFO		
	USER INFO	<b>→</b>		<b>→</b>	INFO(USR)		
				+	200 OK INFO		
	USER INFO	+	· · · · · · · · · · · · · · · · · · ·	+	INFO(USR)		
				→	200 OK INFO		
		Co	ommunicat	tion			
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	RELEASE	+		+	200 OK BYE		
	REL_COM	<b>→</b>					

TP706203	SIP reference	e: RFC 3	261 [4]		ISUP reference: Q.1912.5 [1], clauses 1.3.5.2.5.2.2, 1.3.5.2.2-5.2/ Q.737 [i.10]
TSS reference	ISDN-(ISUP)-SIP/SS	/UUS3			
SIP selection criteria					
ISUP selection criteria					
Test purpose		vice can sted faci	be rejected the lity not imple	with a	REL having the Cause value #29 "facility ed", either with diagnostics (user-to-user
SIP parameter values					
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	+		+	INVITE(IAM UU2 ess)
	RELEASE	<b>→</b>		<b>→</b>	500 Server Internal error(REL#29, 69)
	REL_COM	+		+	ACK
NOTE: The netw	ork or the called user of	cannot su	apport the se	ervice.	

TP706204	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], ses 1.3.5.2.1.1.2, 1.3.5.2.2-5.1/ Q.737 [i.10]	
TSS reference	ISDN-(ISUP)-SIP/SS/UUS	3				
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	UUS3 explicit non-essential - request and acceptance during the active phase of to					
					an UUS3 explicit non-essential	
					neter set to "user-to-user service"	
					et to "request, not essential".  B explicit non-essential request with	
	a FAA having the facility i	indicator	narameter set t	000 n "11	ser-to-user service" and the	
					er set to "service provided".	
SIP parameter	Contract of Hold III and Good	10 4001 11	naioatoro parar		or correct provided :	
values						
ISDN parameter						
values						
Comments	ISDN		SUT		SIP	
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)	
	ALERTING	+		<b>←</b>	180 Ringing(ACM)	
	CONN	+		<b>←</b>	200 OK INVITE	
		C	ommunication			
	FAC(UU3 req not ess)	<b>→</b>		<b>→</b>	INFO(FAR req UU3 not ess)	
				<b>←</b>	200 OK INFO	
	FAC(UU3 ret res)	+		<b>←</b>	INFO(FAR resp UU3 prov)	
	USER INFO	<b>→</b>		<b>→</b>	INFO(USR)	
				<b>←</b>	200 OK INFO	
	USER INFO	+		<b>←</b>	INFO(USR)	
				<b>→</b>	200 OK INFO	
	USER INFO	<b>→</b>		<b>→</b>	INFO(USR)	
				<b>+</b>	200 OK INFO	
	USER INFO	+		<del>(</del>	INFO(USR)	
				<b>→</b>	200 OK INFO	
		C	ommunication			
	DISC	<b>→</b>		<b>→</b>	BYE(REL)	
	RELEASE	+		<b>←</b>	200 OK BYE	
	REL_COM	<b>→</b>				

TP706205	SIP reference: RI	FC 326	1 [4]	clau	ISUP reference: Q.1912.5 [1], se 1.3.5.2.5.2.2/Q.737 [i.10]				
TSS reference	ISDN-(ISUP)-SIP/SS/UUS	S3							
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	FRJ) To verify that the UUS3 e	UUS3 explicit non-essential - explicit rejection during call (service not provided - in FRJ)  To verify that the UUS3 explicit non-essential service can be rejected during the active							
	phase of the call and the to "service 3 not provided		3 field in the	user-to-us	ser indicators in the FRJ are set				
SIP parameter values									
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	→		<b>→</b>	INVITE(IAM)				
	ALERTING	+		+	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE				
			Communica	ation					
	FAC(UU3 req not ess)	→		<b>→</b>	INFO(FAR req UU3 not ess)				
				+	200 OK INFO				
	FAC(UU3 ret err)	+		+	INFO(FRJ resp UU3 not prov)				
				<b>→</b>	200 OK INFO				
			Communica	ation					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	+		+	200 OK BYE				
	REL_COM	→							

TP706206	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 1.3.5.2.5.2.2/Q.737 [i.10]			
TSS reference	ISDN-(ISUP)-SIP/SS/UUS	3		ı			
SIP selection criteria							
ISUP selection criteria	PICS 11/3						
Test purpose	UUS3 explicit non-essential - implicit rejection during call (no indication - discard FAA or FRJ)  To verify that the IUT can successfully complete a call with an UUS3 request in the FAR, if the FAA or FRJ are discarded.						
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE		
		С	ommunicati	on			
	FAC(UU3 req not ess)	<b>→</b>		<b>→</b>	INFO(FAR req UU3 not ess)		
				+	200 OK INFO		
		С	ommunicati	on			
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	RELEASE	+		+	200 OK BYE		
	REL_COM	<b>→</b>					

# A.1.1.2.7 Closed User Group (CUG)

TP707001	SIP reference: RFC 326	1 [4]		ISUP reference:
			clause	Q.1912.5 [1], 1.5.2.1.1 i) a)/Q.735 [i.9]
TSS reference	ISDN-(ISUP)-SIP/SS/CUG		3.0.00	
SIP selection				
criteria				
ISUP selection criteria				
Test purpose	CUG without outgoing access			
	To verify that the IUT can succes			
	interlock code together with an			
	the optional forward call indica			
CID managed an	forward call indicators in the IA	AIVI Should be	set to 150P i	equired all the way.
SIP parameter values				
ISDN parameter				
values				
Comments	ISDN	SU	JT	SIP
	SETUP -	<b>&gt;</b>	<b>→</b>	INVITE(IAM, CUG -OA)
	ALERTING	-	+	180 Ringing
	CONN	+	+	200 OK INVITE
			<b>→</b>	ACK
		Commur	nication	
	DISC	<b>→</b>	→	BYE
		_	+	200 OK BYE
	REL_COM	<b>→</b>		

TP707002	SIP referen	ice: RFC 32	61 [4]	ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]		
TSS reference	ISDN-(ISUP)-SIP/S	S/CUG				
SIP selection criteria						
ISUP selection criteria						
Test purpose	CUG call without of activated To verify that the IU		•		er CUG without IA, no ICB	
SIP parameter values			•			
ISDN parameter values						
Comments	ISDN		SUT		SIP	
	SETUP	+		+	INVITE(IAM, CUG -OA)	
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)	
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)	
			Communicatio	n		
	DISC	+		+	BYE(REL)	
	REL	<b>→</b>		<b>→</b>	200 OK BYE	
	REL_COM	+				

TP707002	SIP reference: F	RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]			
TSS reference	ISDN-(ISUP)-SIP/SS/CU	JG					
SIP selection criteria							
ISUP selection criteria							
Test purpose	CUG call without outgoing access; class of called user CUG without IA, ICB activated  To verify that the IUT rejects the CUG call with a 500 Server Internal Error encapsulated the REL cause #55 "Incoming calls barred within CUG".						
SIP parameter values		-					
ISDN parameter values							
Comments	ISDN	SUT		SIP			
			+	INVITE(IAM, CUG -OA)			
			<b>→</b>	500 Server Internal Error(REL#55)			
			+	ACK			

TP707003	SIP referen	ce: RFC 32	61 [4]	clause	ISUP reference: Q.1912.5 [1], 1.5.2.5.1; table 1-2/Q.735 [i.9]			
TSS reference	ISDN-(ISUP)-SIP/S	S/CUG						
SIP selection criteria								
ISUP selection criteria								
Test purpose	activated	CUG call without outgoing access; class of called user CUG with IA and no ICB activated  To verify that the IUT can successfully establish a CUG call.						
SIP parameter values			·					
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	+		+	INVITE(IAM, CUG -OA)			
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)			
	CONN	→		<b>→</b>	200 OK INVITE(ANM)			
			Communication	on				
	DISC	+		+	BYE(REL)			
	REL	<b>→</b>		<b>→</b>	200 OK BYE			
	REL_COM	+						

TP707004	SIP referer	nce: RFC 326	1 [4]	clause	ISUP reference: Q.1912.5 [1], 1.5.2.5.1; table 1-2/Q.735 [i.9]
TSS reference	ISDN-(ISUP)-SIP/S	S/CUG			
SIP selection criteria					
ISUP selection criteria					
Test purpose	activated		,		CUG with IA and no ICB all with outgoing access.
SIP parameter values			•		<u> </u>
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	+		+	INVITE(IAM, CUG +OA)
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)
			Communicatio	n	, ,
	DISC	+		+	BYE(REL)
	REL	<b>→</b>		<b>→</b>	200 OK BYE
	REL_COM	+			

TP707005	SIP reference	e: RFC	3261 [4]		ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]		
TSS reference	ISDN-(ISUP)-SIP/SS	S/CUG		•	= =		
SIP selection criteria							
ISUP selection criteria							
Test purpose	CUG call without outgoing access; class of called user CUG with IA and ICB activated  To verify that the IUT rejects the CUG call with a 500 Server Internal Error encapsulated the REL cause #55 "Incoming calls barred within CUG".						
SIP parameter values			_				
ISDN parameter values							
Comments	ISDN		SUT		SIP		
				+	INVITE(IAM, CUG -OA)		
				<b>→</b>	500 Server Internal Error(REL#55)		
				+	ACK		

TP707006	SIP reference: F	RFC 3261 [4]		ISUP reference: Q.1912.5 [1], Q.730 [i.1]			
TSS reference	ISDN-(ISUP)-SIP/SS/CU	JG					
SIP selection criteria							
ISUP selection criteria							
Test purpose	CUG call without outgoing access; class of called user non-CUG  To verify that the IUT rejects the CUG call with a 500 Server Internal Error encapsulated the REL cause #87 "User not member of CUG".						
SIP parameter							
values							
ISDN parameter							
values							
Comments	ISDN	SUT		SIP			
			+	INVITE(IAM CUG -OA)			
			<b>→</b>	500 Server Internal Error(REL#87)			
			+	ACK			

TP707007	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]			
TSS reference	ISDN-(ISUP)-SIP/SS/CUG						
SIP selection criteria							
ISUP selection criteria							
Test purpose	CUG call with outgoing access; class of called user non-CUG To verify that the IUT can successfully establish a non-CUG call.						
SIP parameter values							
ISDN parameter							
values							
Comments	ISDN		SUT		SIP-I		
	SETUP	+		+	INVITE(IAM CUG, +OA)		
	ALERTING	→		<b>→</b>	180 Ringing(ACM)		
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)		
				<b>←</b>	ACK		
	Communication						
	DISC	+		+	BYE(REL)		
	REL	<b>→</b>		<b>→</b>	200 OK BYE(RLC)		
	REL_COM	+					

TP707008	SIP reference: F	RFC 3261 [4]	cla	ISUP reference: Q.1912.5 [1], ause 1.5.2.5.1; table 1-2 /Q.735 [i.9]		
TSS reference	ISDN-(ISUP)-SIP/SS/CUG					
SIP selection criteria						
ISUP selection criteria						
Test purpose	Non-CUG call; class of called user CUG without IA  To verify that the IUT rejects the CUG call with a 500 Server Internal Error encapsulated the REL cause # 87 " User not member of CUG".					
SIP parameter values						
ISDN parameter values						
Comments	ISDN (CUG -IA)	SUT		SIP		
			+	INVITE(IAM)		
			<b>→</b>	500 Server Internal Error(REL#87)		
			+	ACK		

TP707009	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]				
TSS reference	ISDN-(ISUP)-SIP/SS/CU	JG						
SIP selection criteria								
ISUP selection criteria								
Test purpose	-	Non-CUG call; class of called user CUG with IA To verify that the IUT can successfully establish a non-CUG call.						
SIP parameter values								
ISDN parameter values								
Comments	ISDN (CUG +IA)		SUT		SIP-I			
	SETUP	+		+	INVITE(IAM)			
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)			
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)			
				+	ACK			
		C	Communication					
	DISC	+		+	BYE(REL)			
	REL	<b>→</b>		<b>→</b>	200 OK BYE(RLC)			
	REL_COM	+						

TP707010	SIP reference: I	RFC 3261 [4]	cla	ISUP reference: Q.1912.5 [1], ause 1.5.2.5.1; table 1-2/Q.735 [i.9]
TSS reference	ISDN-(ISUP)-SIP/SS/CU	JG		-
SIP selection criteria				
ISUP selection criteria				
Test purpose		ects the CUG call wi	th a 50	ed user other CUG without IA 0 Server Internal Error encapsulated
SIP parameter values				
ISDN parameter values				
Comments	ISDN (CUG -IA)	SUT		SIP-I
			+	INVITE(IAM CUG -OA)
			<b>→</b>	500 Server Internal Error(REL#87)
			+	ACK

TP707011	SIP reference: F	RFC 3261 [4]	cla	ISUP reference: Q.1912.5 [1], ause 1.5.2.5.1; table 1-2/Q.735 [i.9]
TSS reference	ISDN-(ISUP)-SIP/SS/CU	JG		
SIP selection criteria				
ISUP selection criteria				
Test purpose		ects the CUG call wi	th a 50	ser other CUG without IA 0 Server Internal Error encapsulated
SIP parameter values				
ISDN parameter values				
Comments	ISDN (CUG -IA)	SUT		SIP-I
			+	INVITE(IAM CUG +OA)
			<b>→</b>	500 Server Internal Error(REL#87)
			+	ACK

TP707012	SIP reference:	RFC 3261 [4]	cla	ISUP reference: Q.1912.5 [1], ause 1.5.2.5.1; table 1-2/Q.735 [i.9]
TSS reference	ISDN-(ISUP)-SIP/SS/CI	UG		
SIP selection criteria				
ISUP selection criteria				
Test purpose	CUG call without outgoing To verify that the IUT returned the REL cause #87 "Us	jects the CUG call wit	th a 50	ser: other CUG with IA 0 Server Internal Error encapsulated
SIP parameter values				
ISDN parameter values				
Comments	ISDN (CUG A +IA)	SUT		SIP-I
			+	INVITE(IAM CUG B -OA)
			<b>→</b>	500 Server Internal Error(REL#87)
			+	ACK

TP707013	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1; table 1-2/Q.735 [i.9]		
TSS reference	ISDN-(ISUP)-SIP/SS/C	UG				
SIP selection criteria						
ISUP selection criteria						
Test purpose	CUG call with outgoing To verify that the IUT of					
SIP parameter			-			
values						
ISDN parameter values						
Comments	ISDN (CUG A +IA)		SUT		SIP-I	
	SETUP	+		+	INVITE(IAM, CUG B +OA)	
	ALERTING	→		<b>→</b>	180 Ringing(ACM)	
	CONN	→		<b>→</b>	200 OK INVITE(ANM)	
			Communicatio	n		
	DISC	+		+	BYE(REL)	
	REL	<b>→</b>		<b>→</b>	200 OK BYE	
	REL_COM	+				

## A.1.1.2.8 SUB-addressing (SUB)

TP708001	SIP reference: RFC 32	261 [4]		ISUP reference: Q.1912.5 [1], e 8.5.2.1.1/Q.731 [i.2]				
TSS reference	ISDN-(ISUP)-SIP/SS/SUB							
SIP selection criteria								
ISUP selection criteria								
Test purpose		Sending the called sub-address in the access transport parameter  To verify that the IUT can include the called sub-address in the access transport  parameter in the IAM						
SIP parameter values	INVITE: IAM encapsulated in t	he MIME body	ATP with called	d sub-address included				
ISDN parameter values	SETUP: called sub-address inc	cluded						
Comments	ISDN	SU	JT	SIP-I				
	SETUP(SUB)	<b>→</b>	<b>→</b>	INVITE(IAM, ATP(SUB))				
	ALERTING	+	+	180 Ringing				
	CONN	+	<del>-</del>	200 OK INVITE				
			→	ACK				
		Commur	nication					
	DISC	<b>→</b>	→	BYE				
	REL	+	<del>-</del>	200 OK BYE				
	REL_COM	<b>→</b>						

TP708002	SIP reference: RF	C 3261 [4]	1	claı	ISUP reference: Q.1912.5 [1], ise 8.5.2.5.1/Q.731 [i.2]		
TSS reference	ISDN-(ISUP)-SIP/SS/SUB						
SIP selection criteria							
ISUP selection criteria							
Test purpose		e successf	ully establi	shed if the I	ort parameter AM contains the sub-address in ddress is passed on to the user		
SIP parameter values	INVITE: IAM encapsulated	in the MIN	ME body A	TP with call	ed sub-address included		
ISDN parameter values	SETUP: called sub-addres	s included	l				
Comments	ISDN		SUT		SIP-I		
	SETUP(SUB)	+		+	INVITE(IAM, ATP(SUB))		
	ALERTING	<b>→</b>		→	180 Ringing(ACM)		
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)		
	Communication  DISC ← BYE(REL)						
	REL	<b>→</b>		<b>→</b>	200 OK BYE(RLC)		
	REL_COM	+					

# A.1.1.2.9 Malicious Call Identification (MCID)

TP709001	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1/Q.731.7 [i.3]					
TSS reference	ISDN-(ISUP)-SIP/SS	/MCID							
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	To verify that the IUT set to "MCID request included" and the cal	Successful MCID request To verify that the IUT can successfully reply to an IDR having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included" and the calling party number included.							
SIP parameter	INFO: The encapsula								
values	INFO: The encapsula Calling party number			ponse in	dicator "included" and the				
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
				+	INFO(IDR requested)				
				→	200 OK INFO				
				→	INFO(IRS included)				
				+	200 OK INFO				
	ALERTING	<del>-</del>		+	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE(ANM)				
				→	ACK				
		(	Communication	1					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE				
	REL_COM	<b>→</b>							

TP709002	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1/Q.731.7 [i.3]		
TSS reference	ISDN-(ISUP)-SIP/SS/	MCID				
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Successful MCID red					
	set to "MCID request" included" and the <b>call</b>	by sending an <b>Í</b> ing party numb	RS with MCID er included.	respon	g the MCID request indicator se indicator set to "MCID	
SIP parameter	INFO: The encapsulat					
values				oonse in	dicator "included" and the	
	Calling party number of	of the originating	User			
ISDN parameter values						
Comments	ISDN		SUT		SIP	
	SETUP	+		+	INVITE(IAM)	
				<b>→</b>	INFO(IDR requested)	
				+	200 OK INFO	
				+	INFO(IRS included)	
				<b>→</b>	200 OK INFO	
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)	
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)	
				<b>→</b>	ACK	
		(	Communication	)		
	DISC	+		+	BYE(REL)	
	REL	<b>→</b>		<b>→</b>	200 OK BYE	
	REL_COM	+				

TP709003	SIP reference:	RFC 3261 [4]			SUP reference: Q.1912.5 [1], - 7.5.2.1.1/Q.731.7 [i.3]			
TSS reference	ISDN-(ISUP)-SIP/SS/N	//CID						
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that the IUT where the second that t	Successful MCID request - after ACM To verify that the IUT will accept and reply correctly to an MCID request after ACM has been received. The IUT should reply to an IDR having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included" and the calling party number included (see note).						
SIP parameter values	INFO: The encapsulated IDR contains the MCID Request indicator "requested" INFO: The encapsulated IRS contains the MCID response indicator "included" and the Calling party number of the originating User							
ISDN parameter values								
Comments	ISDN		SUT		SIP			
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)			
	ALERTING	+		+	180 Ringing(ACM)			
				+	INFO(IDR requested)			
				<b>→</b>	200 OK INFO			
				<b>→</b>	INFO(IRS included)			
				+	200 OK INFO			
	CONN	+		+	200 OK INVITE(ANM)			
				<b>→</b>	ACK			
			Communication	1				
	DISC	→		<b>→</b>	BYE(REL)			
	REL	+		+	200 OK BYE			
	REL_COM	→						
NOTE: This situa	ation may occur e.g. if th	e call has been	forwarded bef	ore reac	hing the destination.			

TP709004	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1/Q.731.7 [i.3]			
TSS reference	ISDN-(ISUP)-SIP/SS/M	CID					
SIP selection criteria							
ISUP selection criteria							
Test purpose	Successful MCID request - after ACM To verify that the IUT will accept and reply correctly to an MCID request after ACM has been received. The IUT should reply to an IDR having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included" and the calling party number included (see note).						
SIP parameter values	INFO: The encapsulated IDR contains the MCID Request indicator "requested" INFO: The encapsulated IRS contains the MCID response indicator "included" and the Calling party number of the originating User						
ISDN parameter values							
Comments	ISDN		SUT		SIP		
	SETUP	+		+	INVITE(IAM)		
	ALERTING	→		<b>→</b>	180 Ringing(ACM)		
				→	INFO(IDR requested)		
				+	200 OK INFO		
				+	INFO(IRS included)		
				<b>→</b>	200 OK INFO		
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)		
				+	ACK		
			Communication	1			
	DISC	+		+	BYE(REL)		
	REL	<b>→</b>		<b>→</b>	200 OK BYE		
	REL_COM	+					
NOTE: This situa	ation may occur e.g. if the	call has beer	forwarded bef	ore reac	hing the destination.		

TP709005	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1/Q.731.7 [i.3]					
TSS reference	ISDN-(ISUP)-SIP/SS/	MCID							
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	To verify that the IUT set to "MCID request"	Successful MCID request with calling sub-address To verify that the IUT can successfully reply to an IDR having the MCID request indicator set to "MCID request" by sending an IRS with MCID response indicator set to "MCID included", the calling party number and a calling sub-address in the access transport							
SIP parameter values	INFO: The encapsulated IDR contains the MCID Request indicator "requested" INFO: The encapsulated IRS contains the MCID response indicator "included", the Calling party number and the calling sub-address of the originating User								
ISDN parameter values									
Comments	ISDN		SUT		SIP				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
				<b>←</b>	INFO(IDR requested)				
				<b>→</b>	200 OK INFO				
				<b>→</b>	INFO(IRS included)				
				<b>←</b>	200 OK INFO				
	ALERTING	<b>←</b>		<b>←</b>	180 Ringing(ACM)				
	CONN	+		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
			Communication	n					
	DISC	<b>→</b>		<b>→</b>	BYE(REL)				
	REL	+		+	200 OK BYE				
	REL_COM	→							

TP70906	SIP reference	e: RFC 3261 [4]		-	SUP reference: Q.1912.5 [1], 7.5.2.1.2/Q.731.7 [i.3]
TSS reference	ISDN-(ISUP)-SIP/SS/	/MCID			
SIP selection criteria					
ISUP selection criteria	PICS 9/1				
Test purpose	MCID request - MCII To verify that the IUT indicator set to "MCI	rejects a MCID re		iding a <b>IR</b>	RS with the MCID response
SIP parameter	INFO: The encapsula	ted IDR contains	the MCID Re	quest inc	licator "requested"
values	INFO: The encapsula	ted IRS contains	the MCID res	ponse in	dicator "not included"
ISDN parameter					
values	IODN		OUT		oin
Comments	ISDN		SUT		SIP
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)
				<del>-</del>	INFO(IDR requested)
				<b>→</b>	200 OK INFO
				→	INFO(IRS not included)
				+	200 OK INFO
	ALERTING	+		+	180 Ringing(ACM)
	CONN	+		+	200 OK INVITE(ANM)
				→	ACK
		С	ommunicatio	n	
	DISC	<b>→</b>		<b>→</b>	BYE(REL)
	REL	+		+	200 OK BYE
	REL_COM	<b>→</b>			

TP70907	SIP reference	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], e 7.5.2.1.2/Q.731.7 [i.3]
TSS reference	ISDN-(ISUP)-SIP/SS/	MCID			
SIP selection criteria					
ISUP selection criteria	PICS 9/1				
Test purpose	MCID request - MCID To verify that the IUT indicator set to "MCID	rejects a MCID	request by ser	nding a <b>IF</b>	RS with the MCID response
SIP parameter	INFO: The encapsula	ted IDR contain	s the MCID Re	quest ind	dicator "requested"
values	INFO: The encapsula	ted IRS contain	s the MCID res	ponse in	dicator "not included"
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	+		+	INVITE(IAM)
				<b>→</b>	INFO(IDR requested)
				+	200 OK INFO
				+	INFO(IRS not included)
				<b>→</b>	200 OK INFO
	ALERTING	→		<b>→</b>	180 Ringing(ACM)
	CONN	<b>→</b>	<u> </u>	→	200 OK INVITE(ANM)
			<u> </u>	+	ACK
			Communicatio	n	
	DISC	+		+	BYE(REL)
	REL	→		→	200 OK BYE
	REL_COM	+			

TP70908	SIP reference: RFC	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], e 7.5.2.5.2/Q.731.7 [i.3]
TSS reference	ISDN-(ISUP)-SIP/SS/MCID				
SIP selection criteria					
ISUP selection criteria	PICS 5/9				
Test purpose	MCID timer (T39) expiry To verify that call setup is co expiry, after having sent the				<b>S</b> is received within timer T39 or set to "MCID requested".
SIP parameter values	INFO: The encapsulated IDR	contai	ns the MCID Re	quest inc	dicator "requested"
ISDN parameter values					
Comments	ISDN		SUT		SIP
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)
				+	INFO(IDR requested)
				<b>→</b>	200 OK INFO
			T39 expiry		
	ALERTING	+		+	180 Ringing(ACM)
	CONN	+		+	200 OK INVITE(ANM)
				<b>→</b>	ACK
			Communicatio	n	
	DISC	<b>→</b>		<b>→</b>	BYE(REL)
	REL	<b>←</b>		+	200 OK BYE
	REL_COM	<b>→</b>			

## A.1.1.2.10 Conference call (CONF)

TP710001	SIP re	eference: RFC 32	261 [4]		UP reference: Q.1912.5 [1],			
					7], clause 1.5.2.1.1.2			
TSS reference	ISDN-(ISUP)	-SIP/SS/CONF		•				
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose			sfully begin the	conference from	an active call and notify			
		arties correctly.						
					stablished" is received in			
			call at the end to	erminal and chec	k that all network resources			
	are released	(see note).	"0115 050					
SIP parameter	INFO: Conte	nt-Type: application	on/ISUP; CPG e	encapsulated in th	ne MIME body			
values	E40 B : 0	ONE: I						
ISDN parameter		ONF invoke com						
values Comments	IFAC: Beginc	ONF return result	component					
ISDN	SUT		SIP 1		SIP 2			
	301 →	<b>→</b>	INVITE		SIF 2			
SETUP(CRx) ALERTING	<del>7</del>	<del>-</del>						
	<del></del>	<del>-</del>	180 Ringing	_				
CONN		<del></del>	200 OK INVITI					
FAC(BeginCONF_ir		<b>→</b>	INFO(CPG cor	nt est)				
FAC(BeginCONF_rr	·) <del>(</del>	←	200 OK INFO					
			ce communicat	ion	1			
DISC(CRx)	<b>→</b>	→	BYE					
RELEASE	<b>←</b>	<b>←</b>	200 OK BYE					
REL_COMP	<b>→</b>							

TP710002	SIP reference: RFC 3261 [4]				UP reference: Q.1912.5 [1], 7], clause 1.5.2.1.1.2			
TSS reference	ISDN-(ISUP)-SII	P/SS/CONE		Q.734 [I.	7], clause 1.5.2.1.1.2			
SIP selection	13014-(1301-)-311	/55/CON						
criteria								
ISUP selection								
criteria								
Test purpose	to add a confer Establish a confer party to the confine the CPG. The note).	Verify that the IUT can successfully begin the conference from idle state and is able to add a conferee to a conference and notify the implied parties correctly. Establish a conference from ISDN to SIP 1. Established a connection to SIP 2 and add this early to the conference. Notify subscriber at SIP 1 by sending him/her "other party added" in the CPG. The conference is released by call clearing by the served user at IADN (see note)						
SIP parameter	INFO: Content-1	ype: application	on/ISUP; CPG e	encapsulated in t	he MIME body			
values	E40 D : 05:							
ISDN parameter	FAC: BeginCON	IF invoke com	ponent					
values	FAC: BeginCON	IF return resul	t component					
	FAC: addCONF DISC: addCONF							
Comments	DISC. addCON	return resuit	component					
ISDN	SUT		SIP 1		SIP 2			
SETUP(CRx)	→	<b>→</b>	INVITE		0.1. 2			
ALERTING	<del>-</del>	<del>-</del>	180 Ringing					
CONN	+	<del>(</del>	200 OK INVIT	F				
FAC(BeginCONF_ir		<b>→</b>	INFO(CPG co					
FAC(BeginCONF_ri		<del>-</del>	200 OK INFO	ili ootj				
	<u> </u>	_						
SETUP(CRy)	<b>→</b>			<b>→</b>	INVITE			
ALERTING	<del>(</del>			<del>(</del>	180 Ringing			
CONN	+			+	200 OK INVITE			
FAC(AddCONF_inv	) →			<b>→</b>	INFO(CPG conf est)			
DISC(AddCONF_rr,	CRy) ←			<del>(</del>	200 OK INFO			
RELEASE	<b>→</b>	<b>→</b>	INFO(CPG pa	rty add)				
REL_COMP	+	+	200 OK INFO	,				
_	, ,	Conferen	ce communica	tion	•			
DISC(CRx)	<b>→</b>	<b>→</b>	BYE					
RELEASE	+	+	200 OK BYE					
REL_COMP	<b>→</b>							
				<b>→</b>	BYE			
				+	200 OK BYE			
new affect		the generic n	otification indi	cator set to "oth	ld be sent by the IUT to the er party added" to the non-			

TP710003	SIP re	eference: RFC 3	261 [4]		GUP reference: Q.1912.5 [1],			
		Q.734 [i.7], cla						
TSS reference	ISDN-(ISUP)	-SIP/SS/CONF			-			
SIP selection								
criteria								
ISUP selection								
criteria	V 26 41 441							
Test purpose	implied partie Establish a co subscriber at and check that	es correctly.  conference from IS  SIP 1 by sending  at the notification	SDN to SIP 1. Acg him/her "other "isolated" is rec	dd SIP 2 to the oparty added" in eleved in the CP0	conference and notify the conference and notify the CPG. Isolate a conferee G. Reattach the conferee.			
SIP parameter values		nt-Type: applicati						
ISDN parameter	FAC: BeginC	ONF invoke com	ponent					
values	FAC: BeginC FAC: addCO DISC: addCC FAC: isolateC FAC: isolateC FAC reattach	FAC: BeginCONF invoke component FAC: BeginCONF return result component FAC: addCONF invoke component DISC: addCONF return result component FAC: isolateCONF invoke component FAC: isolateCONF return result component FAC: reattachCONF invoke component FAC: reattachCONF return result component						
Comments	•		•					
ISDN	SUT		SIP 1		SIP 2			
SETUP(CRx)	→	<b>→</b>	INVITE					
ALERTING	<b>←</b>	+	180 Ringing					
CONN	+	+	200 OK INVIT	E				
FAC(BeginCONF_in	nv <b>)</b> →	<b>→</b>	INFO(CPG co	nf est)				
FAC(BeginCONF_r	r) <b>←</b>	<b>←</b>	200 OK INFO					
SETUP(CRy)	→			<b>→</b>	INVITE			
ALERTING	+			+	180 Ringing			
CONN	<del>(</del>			+	200 OK INVITE			
FAC(AddCONF_inv				<b>→</b>	INFO(CPG conf est)			
DISC(AddCONF_rr,		_		+	200 OK INFO			
RELEASE	<b>→</b>	<b>→</b>	INFO(CPG pa	rty add)				
REL_COMP	<b>←</b>	<b> </b>	200 OK INFO					
540/L 10 1 1	\D\\   \\	Conferen	ce communicat		Investore : "			
FAC(IsolConf_inv,C				<b>→</b>	INFO(CPG isol)			
FAC(IsolConf_rr,CF	Rx) <b>←</b>		INFO/CEC	<b>+</b>	200 OK INFO			
		<b>→</b>	INFO(CPG pa	rty ISOI)				
		Private communi	200 OK INFO	uith OID 4				
EAC/Boot/Oracle	(CD <sub>11</sub> )	Private commu	nication ISDN v		INFO/ODO (1)			
FAC(ReattConf_inv				<b>→</b>	INFO(CPG reatt)			
FAC(ReattConf_rr,0	CRx) <del>(</del>		INFO/ODO		200 OK INFO			
		<b>→</b>	INFO(CPG pa	пу геап)	-			
		Conform	200 OK INFO	lion				
DISC(CD)			ce communicat	lion	1			
DISC(CRx)	<del>)</del>	<b>→</b>						
RELEASE	<b>←</b>	+	200 OK BYE					
REL_COMP	7							
				<b>→</b>	BYE			
NOTE: The second				<b>←</b>	200 OK BYE			
to the affe be sent to	ected conferee the non-affec	and the <b>generic</b> ted conferees. T	notification inches event indicate	dicator set to "or or in the CPG sh	should be sent by the IUT ther party isolated" should ould be set to "progress". the conference.			

TP710004	SIP r	eference:	RFC 3	261 [4]	IS	SUP reference:	
					Q.1912.5 [1], Q.734 [i.7], clause 1.5.2.1.1.5		
TSS reference:	ISDN-(ISUP)-SIP/SS/CONF				Q.734 [I	./], clause 1.5.2.1.1.5	
SIP selection	13DN-(13UP)	-317/33/0	JINF				
criteria							
ISUP selection							
criteria							
Test purpose	Verify that th	e IUT can o	create	a private comm	nunication betw	een the served user and one	
				implied parties			
SIP parameter				on/ISUP; CPG e			
values		,, ,			·	•	
ISDN parameter	FAC: BeginC						
values	FAC: BeginC						
	FAC: addCO						
	DISC: addC0						
	SETUP: split						
	CONNECT: 9	splitCONF	return	result componer	nt		
Comments ISDN	101	· <del>-</del>		OID 4	1	loip o	
	SU	<i>)</i>		SIP 1		SIP 2	
SETUP(CRx) ALERTING	<b>→</b>		<b>→</b>	INVITE			
			<del>-</del>	180 Ringing	_		
CONN	nv) →		<del>-</del>	200 OK INVIT			
FAC(BeginCONF_i			<b>→</b>	INFO(CPG co	nr est)		
FAC( <b>BeginCONF</b> _r	r) <del>–</del>		+	200 OK INFO			
CETUD/CDv/	<b>→</b>			+	<b>→</b>	INVITE	
SETUP(CRy) ALERTING	<del>7</del>		-		<del>7</del>		
CONN	+		-		<del>-</del>	180 Ringing 200 OK INVITE	
			-		<b>→</b>		
FAC(AddCONF_inv DISC(AddCONF_rr	. ,,		-		<del>7</del>	INFO(CPG conf est) 200 OK INFO	
RELEASE	,CRy) <b>←</b>		<b>→</b>	INFO/CDC po		200 OK INFO	
REL_COMP	<del></del>		<del>-</del>	INFO(CPG pa	rty add)		
REL_COMP	<b>\</b>	Cor		ce communicat	lion		
SETUP( <b>SplitConf</b> _i	inv,CRy) →	COI	Heren			INFO(CPG conf disc)	
CONN(SplitIConf					<del>-</del>	200 OK INFO	
COMM(Spinicom_i	i,City)		<b>→</b>	INFO(CPG pa		200 OK INFO	
			<del>-</del>	200 OK INFO	rty Spiit)		
		Private co		nication ISDN v	with SIP 1	1	
FAC( <b>AddCONF</b> _inv	,CRy <b>)</b> →	. IIvale Cl	,,,,,,,,u		<u> →</u>	INFO(CPG conf est)	
DISC(AddCONF_rr					<del>-</del>	200 OK INFO	
RELEASE	,orty) <b>→</b>		<b>→</b>	INFO(CPG pa		200 010 1111 0	
REL_COMP		<del>′</del>	200 OK INFO	,,			
		Cor		ce communicat	tion	I	
DISC(CRx)	<b>→</b>		<del></del>	BYE			
RELEASE	<del>-</del>		<del>-</del>	200 OK BYE			
REL_COMP	<b>→</b>		+				
				1	<b>→</b>	BYE	
				1	<del>-</del>	200 OK BYE	
NOTE 1: The gene	eric notification	n indicato	r set to	n "conference di		ould be sent by the IUT to	

NOTE 1: The **generic notification indicator** set to "conference disconnected" should be sent by the IUT to the affected conferee and the **generic notification indicator** set to "other party split" should be sent to the non-affected conferees. The event indicator in the **CPG** should be set to "progress". The non--affected conferees should not be able to participate in the communication of the private communication.

NOTE 2: See also figure 1-5/ITU-T Recommendation Q.734 [i.7].

TP710005	SIP refere	ence: RFC 3	261 [4]		GUP reference: Q.1912.5 [1], .7], clause 1.5.2.1.1.6
TSS reference	ISDN-(ISUP)-SIP/	SS/CONF		Q.70+[1	., , , , , , , , , , , , , , , , , , ,
SIP selection	10014 (1001 ) 011 /	00,00141			
criteria					
ISUP selection					
criteria					
Test purpose	To verify that IUT	can success	sfully disconnect	a conferee from	the conference, if
	subscriber at SIP	ence from 19 1 by sending	SDN to SIP 1. A g him/her "other	dd SIP 2 to the operty added" in	orrectly. conference and notify the CPG. Release the aring by the served user at
SIP parameter	INFO: Content-Ty	ne: applicati	ion/ISUP: CPG 6	encapsulated in	the MIMF body
values		,			· · · · · · · · · · · · · · · · · · ·
ISDN parameter	FAC: BeginCONF	invoke com	ponent		
values	FAC: BeginCONF				
	FAC: addCONF ir				
	DISC: addCONF				
	FAC: dropCONF i				
	FAC: dropCONF	return result	component		
Comments			_	•	
ISDN	SUT		SIP 1		SIP 2
SETUP(CRx)	<b>→</b>	→	INVITE		
ALERTING	+	<b>←</b>	180 Ringing		
CONN	<del>(</del>	<b>←</b>	200 OK INVIT		
FAC(BeginCONF_		<b>→</b>	INFO(CPG co	nf est)	
FAC( <b>BeginCONF</b> _	rr) <del>(</del>	+	200 OK INFO		
SETUP(CRy)	<b>→</b>			<b>→</b>	INVITE
ALERTING	<del>(</del>			<del>(</del>	180 Ringing
CONN	<b>+</b>			<del>(</del>	200 OK INVITE
FAC(AddCONF_in				<b>→</b>	INFO(CPG conf est)
DISC(AddCONF_r				<b>+</b>	200 OK INFO
RELEASE	<b>→</b>	<b>→</b>	INFO(CPG pa	rty add)	
REL_COMP	+	+	200 OK INFO		
EAC/D CONE:	00 )	Conferen	ce communica		lp. (5
FAC(DropCONF_ir				<b>→</b>	BYE OUR DIVE
FAC( <b>DropCONF</b> _r	r,CRx <b>) ←</b>		INITO/ODO		200 OK BYE
		<b>→</b>	INFO(CPG pa	rty disc)	
			200 OK INFO		
DICC(CD-)			mmunication	I	
DISC(CRx)	<b>→</b>	<b>→</b>	BYE		
RELEASE	<b>←</b>	<del>(</del>	200 OK BYE		+
REL_COMP					l call release procedures i e

NOTE: The IUT should release the leg towards the conferee according to normal call release procedures, i.e. send a **REL** to a conferee connected to the conference. The **generic notification indicator** set to "other party disconnected" should be sent to the non-affected conferees. The event indicator in the **CPG** should be set to "progress".

TP710006	SIP refere	ence: RFC 3	261 [4]	IS	SUP reference: Q.1912.5 [1],
				Q.734 [i	.7], clause 1.5.2.1.1.7
TSS reference	ISDN-(ISUP)-SIP	/SS/CONF		Q., 0 . [.	,
SIP selection	10211 (1001 ) 011 /	00,00111			
criteria					
ISUP selection					
criteria					
Test purpose	requested by the Establish a confer subscriber at SIP	e conferee, a rence from IS 1 by sending	and notify the in SDN to SIP 1. Acg him/her "other	mplied parties odd SIP 2 to the operty added" in	from the conference, if correctly conference and notify the CPG. Release request learing by the served user at
SIP parameter	INFO: Content-Ty	pe: applicati	on/ISUP: CPG e	encapsulated in	the MIME body
values	,		,		,
ISDN parameter	FAC: BeginCONF	invoke com	ponent		
values	FAC: BeginCONF FAC: addCONF in DISC: addCONF FAC: partyDISC in	nvoke compo return result	onent component		
İ	FAC: partyDISC r				
Comments	I AC. partybloc i	etairi resuit t	component		
ISDN	SUT		SIP 1		SIP 2
SETUP(CRx)	→	<b>→</b>	INVITE		011 2
ALERTING	<del>(</del>	<del>-</del>	180 Ringing		
CONN	<del>(</del>	<del>(</del>	200 OK INVIT	E	
FAC(BeginCONF_		→	INFO(CPG co		
FAC( <b>BeginCONF</b> _		+	200 OK INFO	/	
	,				
SETUP(CRy)	<b>→</b>			<b>→</b>	INVITE
ALERTING	<del>(</del>			+	180 Ringing
CONN	+			+	200 OK INVITE
FAC(AddCONF_in	v,CRy <b>)</b> →			<b>→</b>	INFO(CPG conf est)
DISC(AddCONF_r	r,CRy <b>) ←</b>			+	200 OK INFO
RELEASE	<b>→</b>	→	INFO(CPG pa	rty add)	
REL_COMP	+	+	200 OK INFO		
		Conferen	ce communica	tion	
FAC( <b>PartyDisc_</b> inv		+	BYE		
FAC(PartyDisc_rr,	CRy) →	→	200 OK BYE		
				<b>→</b>	INFO(CPG party disc)
				<b>←</b>	200 OK INFO
		Cor	nmunication		
DISC(CRx)	<b>→</b>			<b>→</b>	BYE
RELEASE	+			<b>←</b>	200 OK BYE
REL_COMP NOTE: The IUT	<b>→</b>				l call release procedures, i.e.

NOTE: The IUT should release the leg towards the conferee according to normal call release procedures, i.e. send a RLC in response to the REL to a conferee connected to the conference through ISUP. The generic notification indicator set to "other party disconnected" should be sent to the non-affected conferees. The event indicator in the CPG should be set to "progress".

TP710007	SIP r	eference: RF	C 32	261 [4]			UP reference: Q.1912.5 [1],
					Q 73		7], clause 1.5.2.1.1.8
TSS reference	ISDN-(ISUP)	)-SIP/SS/CON	JF		ασ	<u> </u>	71, 014400 11012111110
SIP selection	10211 (1001)	,,,	•				
criteria							
ISUP selection							
criteria							
Test purpose	requested by conferee. Establish a consubscriber at	To verify that IUT can successfully disconnect all conferees from the conference, if equested by the served user, and initiate the normal call release procedure towards each conferee.  Establish a conference from ISDN to SIP 1. Add SIP 2 to the conference and notify subscriber at SIP 1 by sending him/her "other party added" in the CPG. The conference is					
SIP parameter	INFO: Conto	ent Tunes cont	y une	e served user at on/ISUP; CPG e	noonsulated	in t	). ba MIME bady
values	INFO. Conte	пт-туре. аррг	icali	JII/13UF, CFG 6	ncapsulated	III T	ne wiiw⊏ boay
ISDN parameter	EAC: Bogin	CONF invoke	comi	nonont			
values		CONF return r					
Values		ONF invoke co					
		ONF return re					
Comments			-				
ISDN	SU	JT		SIP 1			SIP 2
SETUP(CRx)	<b>→</b>		<b>→</b>	INVITE			-
ALERTING	<del>(</del>		+	180 Ringing			
CONN	+		+	200 OK INVIT	E		
FAC(BeginCONF_ir	ıv) →		<b>→</b>	INFO(CPG cor			
FAC( <b>BeginCONF</b> _rr			+	200 OK INFO			
	,						
SETUP(CRy)	<b>→</b>					<b>→</b>	INVITE
ALERTING	+					<del>(</del>	180 Ringing
CONN	+					<del>(</del>	200 OK INVITE
FAC(AddCONF_inv,	.CRv) →					<b>→</b>	INFO(CPG conf est)
DISC(AddCONF_rr,						<del>(</del>	200 OK INFO
RELEASE	→		<b>→</b>	INFO(CPG pa	rtv add)		
REL_COMP	+		<del>-</del>	200 OK INFO	,,		
	I .	Confe	rend	ce communicat	ion		
DISC(CRx)	→		<b>→</b>	BYE			
RELEASE			200 OK BYE				
REL_COMP	<b>→</b>					<b>→</b>	INFO(CPG party disc)
	-					<u>-</u>	200 OK INFO
						<u>-</u>	BYE
						<u>-</u>	200 OK BYE
NOTE: The IUT s	hould send R	EL to all confe	eree	s connected to t			1

TP710008	SIP ref	ference: RFC 3	261 [4]	ISUP reference: Q.1912.5 [1], Q.734 [i.7], clause 1.6.15		
TSS reference	ISDN-(ISUP)-S	SIP/SS/CONF				
SIP selection						
criteria						
ISUP selection						
criteria						
Test purpose	Verify that no r	retrieve notificat	tion is sent to a ι	user put on hold	and subsequently added to	
		call, but that the	IUT sends the "	conference esta	blished" notification to the	
	held user.					
SIP parameter values				encapsulated in t	he MIME body	
ISDN parameter		NF invoke com				
values		NF return resul				
		F invoke compo				
	DISC: addCO	NF return result	component			
Comments			_			
ISDN	SUT		SIP 1		SIP 2	
SETUP(CRx)	<b>→</b>	<b>→</b>	INVITE			
ALERTING	<b>←</b>	+	180 Ringing			
CONN	+	+	200 OK INVIT			
FAC( <b>BeginCONF</b> _ir		<b>→</b>	INFO(CPG co	nf est)		
FAC(BeginCONF_r	r) <del>(</del>	+	200 OK INFO			
SETUP(CRy)	→			→	INVITE	
ALERTING	+			<del>(</del>	180 Ringing	
CONN	<del>(</del>			<del>(</del>	200 OK INVITE	
HOLD	<b>→</b>			<b>→</b>	INFO(CPG hold)	
				+	200 OK INFO	
FAC(AddCONF_inv				<b>→</b>	INFO(CPG conf est)	
DISC(AddCONF_rr,				+	200 OK INFO	
RELEASE	<b>→</b>	<b>→</b>	INFO(CPG pa	rty add)		
			200 OK INFO			
			ce communica	tion		
DISC(CRx)	<b>→</b>	<b>→</b>	BYE			
RELEASE	+	+	200 OK BYE			
REL_COMP	<b>→</b>			<b>→</b>	INFO(CPG party disc)	
				+	200 OK INFO	
				<b>→</b>	BYE	
				←	200 OK BYE	

TP710009	SIP referer	SIP reference: RFC 3261 [4]			GUP reference: Q.1912.5 [1], [i.7], clause 1.6.15
TSS reference	ISDN-(ISUP)-SIP/S	S/CONF			
SIP selection					
criteria					
ISUP selection					
criteria					
Test purpose					conferees when the
	conference controll	er puts the	conference on I	nold.	
SIP parameter	INFO: Content-Typ	e: applicati	on/ISUP; CPG e	encapsulated in	the MIME body
values					
ISDN parameter	FAC: BeginCONF i				
values	FAC: BeginCONF				
	FAC: addCONF inv				
Cammanta	DISC: addCONF re	turn result	component		
Comments ISDN	SUT		CID 4	I	SIP 2
			SIP 1		SIP 2
SETUP(CRx)	<b>→</b>	<b>→</b>	INVITE		
ALERTING	+	<del>-</del>	180 Ringing	_	
CONN	<b>←</b>	<del>(</del>	200 OK INVIT		
FAC(BeginCONF_		<b>→</b>	INFO(CPG co	nf est)	
FAC(BeginCONF_	rr) <del>(</del>	+	200 OK INFO		
057110(00.)					N 075
SETUP(CRy)	<b>→</b>			<b>→</b>	INVITE
ALERTING	+			<del>(</del>	180 Ringing
CONN	<del>(</del>			<del>(</del>	200 OK INVITE
FAC( <b>AddCONF</b> _in				<b>→</b>	INFO(CPG conf est)
DISC(AddCONF_r				<b>+</b>	200 OK INFO
RELEASE	<b>→</b>	<b>→</b>	INFO(CPG pa	rty add)	
REL_COMP	<b>←</b>	<b>←</b>	200 OK INFO		
		Conferen	ce communicat	ion	
HOLD	<b>→</b>				
RETRIVE	<b>→</b>				
DISC(CRx)	<b>→</b>				
RELEASE		<b>←</b> 200 OK BYE			
REL_COMP	→			→	INFO(CPG party disc)
				+	200 OK INFO
				<b>→</b>	BYE
				+	200 OK BYE

# A.1.1.2.11 Explicit Call Transfer (ECT)

TP711001	SIP reference	SIP reference: RFC 3261 [4]						ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.1 a)/Q.732.7 [i.5]			
TSS reference	ISDN-(ISUP)-SIP/SS	/FCT	-			Clause 1.5	.2.1	. 1.11 ajr w. 11 52.11 [1.5]			
SIP selection		,									
criteria											
ISUP selection											
criteria											
Test purpose	transfer number To verify that the IUT number when the ca	Capability of storing and sending the additional calling party number in the call transfer number  To verify that the IUT is able to store the additional calling party number in the generic number when the calling party number and the generic number have been received from the remote user. This information is sent by the IUT to the other remote user in the									
	call transfer numbe	all transfer number in either the FAC or CPG when the call transfer is activated.									
SIP parameter values	INVITE: encapsulate INVITE B SDP sendo INFO C: encapsulate number derive	NVITE: encapsulated IAM contains the additional calling party number of user B  NVITE B SDP sendonly, encapsulated CPG generic notification remote hold  NFO C: encapsulated FAC contains generic notification call transfer active, call transfer number derived from the additional calling party number of user B (SIP-I 1)									
ISDN parameter		AC: ECT invoke request component									
values	DISCONNECT: ECT invoke return result component										
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3			
	SETUP	<b>←</b>		+	INVITE(	IAM)					
	ALERTING	<b>→</b>				ging(ACM)					
	CONN	<b>→</b>				INVITE(ANM)					
				+	ACK						
	HOLD	<b>→</b>		<b>^</b>	INVITE(	CPG hold)					
					200 OK	INVITE					
				<b>→</b>	ACK						
	SETUP	<b>→</b>					<b>-</b>	INVITE(IAM)			
	ALERTING	+						180 Ringing(ACM)			
	CONN	+						200 OK INVITE(ANM)			
	001111	† <u> </u>						ACK			
	FAC(ECT invoke)	<b>→</b>						7.5.1			
	DISCONNECT(rr)	<del>-</del>		<b>→</b>	INFO (F	AC ect active)					
	RELEASE	<b>→</b>			200 OK						
	RELEASE COMPL	+			1 2 2		→	INFO (FAC ect active)			
								200 OK INFO			
			•		•		1	<u>-</u>			
						BYE(REL)	<b>→</b>	BYE(REL)			
					20			200 OK BYE(RLC			

TP711002	SIP referenc	SIP reference: RFC 3261 [4]						ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.1 a)/Q.732.7 [i.5]			
TSS reference	ISDN-(ISUP)-SIP/SS	/FCT	-			10050 7.0.	<u> </u>	in ajrenozir [iio]			
SIP selection	10211 (1001 ) 011 / 00	,									
criteria											
ISUP selection											
criteria											
Test purpose	Capability of storing	Capability of storing and sending the calling party number in the call transfer									
	number										
		To verify that the IUT is able to store the calling party number when only this CLI has									
	been received from t										
		all tra	ınsfer nu	mb	<b>er</b> in either the	FAC or C	PG	when the call transfer is			
OID	activated.										
SIP parameter		NVITE: encapsulated IAM contains the calling party number of user B NVITE B SDP sendonly, encapsulated CPG generic notification remote hold									
values	INVITE B SDP sendo	only,	encapsul	ate	a CPG generic	notificatio	n re	emote noid			
		INFO C: encapsulated FAC contains generic notification call transfer active, call transfer									
ISDN parameter	number derived from the calling party number of user B (SIP-I 1)  FAC: ECT invoke request component										
values	DISCONNECT: ECT invoke return result component										
Comments	ISDN 2	111001	SUT	100	SIP-I 1			SIP-I 3			
Comments	SETUP	+	001	4	INVITE(IAM)			013			
	ALERTING	<b>→</b>			180 Ringing(A	(CM)					
	CONN	<b>→</b>			200 OK INVIT						
	001414	Ť			ACK	<u> </u>					
					7.01						
	HOLD	<b>→</b>		<b>→</b>	INVITE(CPG	hold)					
					200 OK INVIT						
					ACK						
					7.0.1						
	SETUP	<b>→</b>					<b>→</b>	INVITE(IAM)			
	ALERTING	+						180 Ringing(ACM)			
	CONN	+						200 OK INVITE(ANM)			
								ACK			
	FAC(ECT invoke)	<b>→</b>						-			
	DISCONNECT(rr)	+		<b>→</b>	INFO (FAC ed	ct active)					
	RELEASE	<b>→</b>			200 OK INFO						
	RELEASE COMPL	+					<b>→</b>	INFO (FAC ect active)			
								200 OK INFO			
			1		·			I .			
					E	BYE(REL)	<b>→</b>	BYE(REL)			
								200 OK BYE(RLC			

TP711003	SIP referenc	SIP reference: RFC 3261 [4]						ISUP reference: Q.1912.5 [1],				
								1.1 b)/Q.732.7 [i.5]				
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT	-									
SIP selection												
criteria												
ISUP selection												
criteria												
Test purpose	Capability of storing	g and	d sendin	g th	ne addit	ional connected	l nu	mber in the call				
	transfer number											
	To verify that the IUT											
	the remote user. This							ave been received from				
SIP parameter		nsfer number in either the FAC or CPG when the call transfer is activated.  VITE B SDP sendonly, encapsulated CPG generic notification remote hold										
values		00 OK INVITE: encapsulated ANM containing the additional connected number										
Tuidos		NFO B: encapsulated FAC contains generic notification call transfer active, call transfer										
		number derived from the additional connected of user C (SIP-I 2)										
ISDN parameter		AC: ECT invoke request component										
values	DISCONNECT: ECT	DISCONNECT: ECT invoke return result component										
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3				
	SETUP	+		<b>←</b>	INVITE	(IAM)						
	ALERTING	<b>→</b>		<b>→</b>	180 Rir	nging(ACM)						
	CONN	<b>→</b>		<b>→</b>	200 OK	(INVITE(ANM)						
				<b>←</b>	ACK							
	HOLD	<b>→</b>		<b>→</b>	INVITE	(CPG hold)						
				<b>←</b>	200 OK	INVITE						
				<b>→</b>	ACK							
	SETUP	<b>→</b>					<b>→</b>	INVITE(IAM)				
	ALERTING	+					+	180 Ringing(ACM)				
	CONN	+					+	200 OK INVITE(ANM)				
							<b>→</b>	ACK				
	FAC(ECT invoke)	<b>→</b>										
	DISCONNECT(rr)	+				FAC ect active)						
	RELEASE	<b>→</b>		<b>←</b>	200 OK	INFO						
	RELEASE COMPL	+						INFO (FAC ect active)				
							<b>←</b>	200 OK INFO				
								BYE(REL)				
					20	00 OK BYE(RLC	+	200 OK BYE(RLC				

TP711004	SIP reference	e: RI	C 3261	[4]		ISUP reference: Q.1912.5 [1],					
						clause 7.5.	2.1.	.1.1 b)/Q.732.7 [i.5]			
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT	•								
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	To verify that the IUT	Capability of storing and sending the connected number in call transfer number To verify that the IUT is able to store connected number when only this COL has been received from the remote user. This information is sent by the IUT to the other remote user									
		n the call transfer number in either the FAC or CPG when the call transfer is activated.									
SIP parameter	INVITE B SDP sendo	NVITE B SDP sendonly, encapsulated CPG generic notification remote hold									
values		200 OK INVITE: encapsulated ANM containing the connected number									
							ınsf	er active, call transfer			
		number derived from the connected of user C (SIP-I 2)									
ISDN parameter	FAC: ECT invoke request component										
values	DISCONNECT: ECT invoke return result component										
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3			
	SETUP	<b>←</b>			INVITE						
	ALERTING	→				nging(ACM)					
	CONN	<b>→</b>				(INVITE(ANM)					
				<b>←</b>	ACK						
	HOLD	<b>→</b>				(CPG hold)					
						( INVITE					
				<b>→</b>	ACK						
	SETUP	<b>→</b>						INVITE(IAM)			
	ALERTING	+						180 Ringing(ACM)			
	CONN	<b>←</b>						200 OK INVITE(ANM)			
							<b>→</b>	ACK			
	FAC(ECT invoke)	<b>→</b>									
	DISCONNECT(rr)	+		<b>→</b>	INFO (	FAC ect active)					
	RELEASE	<b>→</b>		<b>←</b>	200 Ok	(INFO					
	RELEASE COMPL	<b>←</b>	→ INFO (FAC ect ac								
			<b>←</b> 200 OK INFO								
								•			
						BYE(REL)	<b>→</b>	BYE(REL)			
					20	00 OK BYE(RLĆ		200 OK BYE(RLC			

TP711005	SIP reference	e: RF	C 3261 [4]	]		_	GUP reference: Q.1912.5 [1],					
								.1.2.1/Q.732.7 [i.5]				
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT										
SIP selection												
criteria												
ISUP selection criteria												
Test purpose	To verify that the local prevention procedure with call transfer ref	Loop prevention procedure - initiation and successful response  To verify that the local exchange controlling the ECT can successfully initiate the loop prevention procedure by sending LOP with loop prevention indicator set to "request" and with call transfer reference for both calls.										
	To verify that the local exchange controlling the ECT can successfully perform a call transfer if a <b>LOP</b> with <b>loop prevention indicator</b> set to "response" is received and "no loop exists", and the call identity matches the one used by the IUT.											
SIP parameter values	INFO: encapsulated LOP request, call transfer reference INFO: encapsulated LOP response, call transfer reference, response indicator: "no loop exists"											
ISDN parameter values		OAIGG										
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3				
	SETUP	+			INVITE(I	AM)						
	ALERTING	<b>→</b>	7	•	180 Ring	jing(ACM)						
	CONN	<b>→</b>	7	•	200 OK	INVITE(ANM)						
			•	-	ACK							
	HOLD	<b>→</b>				CPG hold)						
					200 OK I	INVITE						
				<b>&gt;</b>	ACK							
	SETUP	<b>→</b>					_	INVITE(IAM)				
	ALERTING	+						180 Ringing(ACM)				
	CONN	+						200 OK INVITE(ANM)				
	COMM	_						ACK				
							7	ACK				
				_	INFO/LC	P request)						
					200 OK I							
					200 01(1	11110	-	INFO(LOP request)				
					1		4	200 OK INFO				
							Ť	200 011111 0				
			•	_	INFO(LC	P response)						
					200 OK I							
						····						
							+	INFO(LOP response)				
								200 OK INFO				
	EAC/ECT: L \	+			<u> </u>		-					
	FAC(ECT invoke)	<b>→</b>		_	INIEO /E	10 and a still 1	-					
	DISCONNECT(rr)	+				AC ect active)						
	RELEASE COMPL	<b>→</b>	•	_	200 OK I	INFU	_	INIEO (EAC ant antiva)				
	RELEASE COMPL	7						INFO (FAC ect active)				
					1		1	200 OK INFO				
						RVE/DEI \		BVE(REL)				
					200							
	BYE(REL) → BYE(REL)  200 OK BYE(RLC ← 200 OK BYE(RLC											

TP711006	SIP reference	FC 3261	[4]		G	can successfully initiate a call transfer "call transfer, active" or "call transfer,					
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT	•								
SIP selection criteria											
ISUP selection criteria											
Test purpose	To verify that the loca by sending <b>FAC</b> with	Facility message with generic notification sent to the remote user  To verify that the local exchange controlling the ECT can successfully initiate a call transfer by sending FAC with the generic notification set to "call transfer, active" or "call transfer, alerting" and the service activation parameter set to "call transfer".									
SIP parameter	INFO B: encapsulate	NFO B: encapsulated FAC contains generic notification call transfer active									
values	INFO C: encapsulated FAC contains generic notification call transfer active										
ISDN parameter	FAC: ECT invoke req	FAC: ECT invoke request component									
values	DISCONNECT: ECT invoke return result component										
Comments	ISDN 2										
	SETUP	+			INVITE						
	ALERTING	<b>→</b>				nging(ACM)					
	CONN	<b>→</b>		<b>^</b>	200 OK	(INVITE(ANM)					
				₩	ACK						
	HOLD	<b>→</b>		<b>→</b>	INVITE	(CPG hold)					
				+	200 OK	INVITE					
				<b>→</b>	ACK						
	SETUP	<b>→</b>					<b>→</b>	INVITE(IAM)			
	ALERTING	+						180 Ringing(ACM)			
	CONN	+						200 OK INVITE(ANM)			
							<b>→</b>	ACK			
	FAC(ECT invoke)	<b>→</b>									
	DISCONNECT(rr)	+		<b>→</b>	INFO (I	FAC ect active)					
	RELEASE	<b>→</b>			200 OK						
	RELEASE COMPL	+					<b>→</b>	INFO (FAC ect active)			
								200 OK INFO			
					•			•			
							<b>→</b>	BYE(REL)			
					20	00 OK BYE(RLC	<b>←</b>	200 OK BYE(RLC			

TP711007	SIP reference	SIP reference: RFC 3261 [4]						ISUP reference:				
								)12.5 [1], I.2.2 a)/Q.732.7 [i.5]				
TSS reference	ISDN-(ISUP)-SIP/SS/	FCT	-			Clause 7.5.2		1.2.2 a)/Q./32./ [1.3]				
SIP selection	10011 (1001 ) 011 7007											
criteria												
ISUP selection												
criteria												
Test purpose	Call progress message with generic notification sent to the remote user To verify that the local exchange (controlling the ECT) can successfully initiate a call transfer by sending CPG with the generic notification set to "call transfer, active" and the											
	service activation pa	ram	with the <b>c</b> leter set t	<b>jen</b> :0 "(	eric not call trans	sfer".	call	transfer, active and the				
SIP parameter	INFO C: encapsulated CPG contains generic notification call transfer active											
values	INFO B: encapsulated	I FA	C contair	าร g	eneric r	notification call tra	nsf	er alerting				
	INFO B encapsulated				eneric n	otification call tra	nsfe	er active				
ISDN parameter		FAC: ECT invoke request component										
values	DISCONNECT: ECT invoke return result component											
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3				
	SETUP	<b>←</b>			INVITE							
	ALERTING	<b>→</b>				nging(ACM)						
	CONN	<b>→</b>				(INVITE(ANM)						
				<b>←</b>	ACK							
		<u> </u>										
	HOLD	<b>→</b>				(CPG hold)						
						INVITE						
				<b>→</b>	ACK							
	SETUP	<b>→</b>					<b>→</b>	INVITE(IAM)				
	ALERTING	+					+	180 Ringing(ACM)				
	FAC(ECT invoke)	<b>→</b>										
	DISCONNECT(rr)	+		<b>→</b>	INFO (I	FAC ect alert)						
	RELEASE	<b>→</b>			200 Ok							
	RELEASE COMPL	+				· · · · · ·	<b>→</b>	INFO (CPG ect active)				
								200 OK INFO				
							+	200 OK INVITE(ANM)				
								ACK				
				<b>→</b>	INFO (I	FAC ect active)						
					200 Ok							
				l		BYE(REL)	4	RVE(REL)				
		<del>                                     </del>			20			200 OK BYE(RLC				
	L	1		l		ON BIE(KLC	_	ZOO ON BTE(NLC				

TP711008						UP	JP reference:			
								912.5 [1],		
						clause 7.5.2	2.1.1	I.2.2 b)/Q.732.7 [i.5]		
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT								
SIP selection										
criteria										
ISUP selection criteria										
Test purpose	call is alerting			•				is invoked while one as soon as the local		
	exchange (controlling remote user the FAC notification set to "c	the E with	ECT) red <b>service</b>	ceiv <b>act</b>	es the A ivation	NM, it can succe	essfi	ully send to the other		
SIP parameter	INFO B encapsulated					otification call tra	nsfe	er active		
values										
ISDN parameter values										
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3		
	SETUP	+		+	INVITE	(IAM)				
	ALERTING	<b>→</b>		<b>→</b>	180 Rii	nging(ACM)				
	CONN	<b>→</b>				(INVITE(ANM)				
				+	ACK	,				
	HOLD	→				(CPG hold)				
						( INVITE				
				→	ACK					
	SETUP	<b>→</b>					<b>→</b>	INVITE(IAM)		
	ALERTING	+					+	180 Ringing(ACM)		
	FAC(ECT invoke)	<b>→</b>		-						
	DISCONNECT(rr)	<del>-</del>		<b>→</b>	INFO (	FAC ect alert)				
	RELEASE	<b>→</b>			200 Or					
	RELEASE COMPL	<del>-</del>		Ì	200 01	(    (   (	<b>→</b>	INFO (CPG ect active)		
	TELEPTOE GOIVII E							200 OK INFO		
								200 OK INVITE(ANM)		
							<b>→</b>	ACK		
						FAC ect active)				
				<b>←</b>	200 Ok	( INFO				
						BYE(REL)		BYE(REL)		
					20	00 OK BYE(RLC	<b>←</b>	200 OK BYE(RLC		

TP711009	SIP reference	SIP reference: RFC 3261 [4]						ISUP reference: Q.1912.5 [1], clause 7.5.2.1.1.2.2 b)/Q.732.7 [i.5]			
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT	-			Clause 7.5.	<b>2.</b> I .	1.2.2 D/Q./32./ [1.3]			
SIP selection	10014-(1001 )-011 /00	,, LO 1									
criteria											
ISUP selection											
criteria											
Test purpose	parameter when the To verify that, in case other remote user up with the information in number and an addi ANM.	e ECT e the oon re receive tional	F is involuted in its investigation in the reconnected in the seconnected in the seconnec	ked Ivol he ge ed	I while o ked while ANM cor eneric nu number i	ne call is alerti one call is aler nveys the call tr imber paramete n the generic n	ng ting, ans er if l uml	fer number parameter both the connected per are received in the			
SIP parameter values	connected number INFO B: encapsulate	INFO B: encapsulated FAC contains generic notification call transfer active and call									
1001	transter number deri	transfer number derived from the additional connected number									
ISDN parameter											
values Comments	ICDN 0		CUT		CID I 4		1	CID I 2			
Comments	ISDN 2	-	SUT	_	SIP-I 1	1000		SIP-I 3			
	SETUP	+		_	INVITE(						
	ALERTING	<b>→</b>				ging(ACM) INVITE(ANM)					
	CONN	7			ACK	INVITE(ANIVI)	-				
				_	ACK		-				
	HOLD	<b>→</b>		_	INDUITE	CDC Fala)	-				
	HOLD	7			200 OK	CPG hold)	-				
					ACK	IINVIIE	-				
				7	ACK		-				
	CETUD						+	INIVITE (IANA)			
	SETUP	<b>→</b>						INVITE(IAM)			
	ALERTING	_					_	180 Ringing(ACM)			
	FAC(FCT involve)	<b>→</b>									
	FAC(ECT invoke)	_		_	INICO /F	· A C a at a la :-+\	+				
	DISCONNECT(rr)	+				AC ect alert)	-				
	RELEASE	<b>→</b>		_	200 OK	INFU		INITO (ODOtti: \			
	RELEASE COMPL	+						INFO (CPG ect active)			
		-					+	200 OK INFO			
		-					+_	000 01/ 100 (777/4011)			
		-						200 OK INVITE(ANM)			
				_	INIEO (E	· ^ · · · · · · · · · · · · · · · · · ·	7	ACK			
		-		_		AC ect active)	-				
		-		_	200 OK	INFU					
					I	חער(סבי ׳	1. \$	DVE(DEL)			
		-			22	BYE(REL)		BYE(REL)			
					20	OK BYE(RLC	<b>←</b>	200 OK BYE(RLC			

TP711010	SIP reference	SIP reference: RFC 3261 [4]						ISUP reference: Q.1912.5 [1],			
								1.2.2 b)/Q.732.7 [i.5]			
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT	•								
SIP selection											
criteria											
ISUP selection criteria											
Test purpose	when the ECT is in	Capability of sending the connected number in the call transfer number parameter when the ECT is invoked while one call is alerting  To verify that, in case the ECT is invoked while one call is alerting, the FAC sent to the									
		other remote user upon receipt of the <b>ANM</b> conveys the <b>call transfer number</b> parameter									
		with the information received in the <b>connected number</b> parameter if only the <b>connected</b>									
		number is received in the ANM.									
SIP parameter	200 OK INVITE: enc			Мс	ontains t	he connected nu	ımbı	er and the additional			
values	connected number	ароаі	a.oa /	0	ornanio t	110 001111001001110		or and the additional			
	INFO B: encapsulate	d FA	C contai	ns c	generic n	otification call tra	ansf	er active and call			
	transfer number deri										
ISDN parameter											
values											
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3			
	SETUP	+			INVITE(						
	ALERTING	<b>→</b>				ging(ACM)					
	CONN	<b>→</b>				INVITE(ANM)					
				<b>←</b>	ACK						
	HOLD	<b>→</b>				(CPG hold)					
					200 OK	INVITE					
				<b>→</b>	ACK						
	SETUP	<b>→</b>						INVITE(IAM)			
	ALERTING	<b>←</b>					+	180 Ringing(ACM)			
	FAC(ECT invoke)	<b>→</b>									
	DISCONNECT(rr)	<b>←</b>				AC ect alert)					
	RELEASE	<b>→</b>		<b>←</b>	200 OK	INFO					
	RELEASE COMPL	<b>←</b>						INFO (CPG ect active)			
							<b>←</b>	200 OK INFO			
				1							
							200 OK INVITE(ANM)				
						→	ACK				
				_		AC ect active)					
			+	200 OK	INFO						
						·					
						BYE(REL)		BYE(REL)			
					20	0 OK BYE(RLC	+	200 OK BYE(RLC			

TP711011	SIP referenc		[4]		ISUP reference: Q.1912.5 [1], clauses 7.3; 7.5.2.3.1/Q.732.7 [i.5]							
TSS reference	ISDN-(ISUP)-SIP/SS	/ECT										
SIP selection												
criteria												
ISUP selection criteria												
Test purpose	To verify that the IUT	Call transfer number - conversion to international number To verify that the IUT converts the call transfer number to international format. The nature of address indicator shall be set to "international number".										
SIP parameter values												
ISDN parameter values												
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3				
	SETUP	1		+	INVITE	(IAM)						
	ALERTING	1		1	180 Rir	nging(ACM)						
	CONN	<b>↓</b>				(INVITE(ANM)						
				+	ACK							
	HOLD	<b>+</b>		<b>→</b>	INVITE	(CPG hold)						
						INVITE						
				<b>→</b>	ACK							
	SETUP	<b>→</b>					_	INVITE(IAM)				
	ALERTING	<del>-</del>		-				180 Ringing(ACM)				
	CONN	+						200 OK INVITE(ANM)				
	CONN	+		-				ACK				
	FAC(ECT invoke)	<b>→</b>		+			<del>                                     </del>	/ IOIX				
	DISCONNECT(rr)	<del>-</del>		4	INFO (	FAC ect active)						
	RELEASE	<b>→</b>			200 OK		<del>                                     </del>					
	RELEASE COMPL	<del>-</del>		+	200 01	C II VI O	<b>→</b>	INFO (FAC ect active)				
	TELLITOL OOM L	+		1				200 OK INFO				
				1	1			200 01(1141 0				
						BYE(REL)	<b>→</b>	BYE(REL)				
					20			200 OK BYE(RLC				

TP711012	SIP reference	SIP reference: RFC 3261 [4]					2.19	reference: 912.5 [1],			
						clauses 7.3	; 7.	5.2.4.1/Q.732.7 [i.5]			
TSS reference	ISDN-(ISUP)-SIP/SS	S/ECT	-								
SIP selection											
criteria											
ISUP selection											
criteria Test purpose	0-11 (	Call transfer number - removal of own country code									
	To verify that the IU number if it is the ne to "national (signification).	To verify that the IUT removes the country code in the address signals of the <b>call transfer number</b> if it is the network's own country code. The nature of address indicator shall be set to "national (significant) number".									
SIP parameter	INVITE SIP-I 1: enca					alling party number	er N	oA "international			
values		number with the networks own country code.									
						ontains connected	d nu	ımber NoA "international			
	number with the net	works	own cou	ıntr	y code.	!! +u-u!	L -	alambia al fuanza a a a a a a a d			
	number NoA "nation			ont	ains the	call transfer num	ber	derived from connected			
	INFO SIP-I 3: encap			ont	aine tha	call transfer num	har	derived from calling			
	party number NoA "r				anis inc	can transfer fluin	DEI	derived from calling			
ISDN parameter	party namber 140/1	and hamber took hamber									
values											
Comments	ISDN 2		SUT		SIP-I 1			SIP-I 3			
	SETUP	+		+	INVITE	(IAM)					
	ALERTING	<b>→</b>		<b>→</b>	180 Rir	nging(ACM)					
	CONN	<b>→</b>		<b>→</b>	200 OK	(INVITE(ANM)					
				<b>←</b>	ACK						
	HOLD	<b>→</b>		<b>→</b>	INVITE	(CPG hold)					
				+	200 Ok	INVITE					
				<b>→</b>	ACK						
	SETUP	<b>→</b>						INVITE(IAM)			
	ALERTING	+						180 Ringing(ACM)			
	CONN	+						200 OK INVITE(ANM)			
							→	ACK			
	FAC(ECT invoke)	<b>→</b>									
	DISCONNECT(rr)	+	-			FAC ect active)					
	RELEASE	<b>→</b>		<b>←</b>	200 Ok	(INFO					
	RELEASE COMPL	+						INFO (FAC ect active)			
							←	200 OK INFO			
			•					1			
								BYE(REL)			
					20	00 OK BYE(RLC	<b>←</b>	200 OK BYE(RLC			

# A.1.1.2.12 Call Diversion (CFB, CFNR, CFU, CD)

TP712001	SIP reference: RFC 3261 [4]	cla	ISUP reference: Q.1912.5 [1], clause 2.5.2.1.1/Q.732 [i.4]						
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV								
SIP selection									
criteria									
ISUP selection									
criteria									
Test purpose	"Call is diverting" indication received in A								
SIP parameter values ISDN parameter values	To verify that a call can be successfully established, if diversion occurs. The encapsulated ACM contains the generic notification indicator set to "call is diverting", the call diversion information and the redirection number.  Applicable redirection reason in the call diversion information: "busy"  CFB(n); CFB(u,l)  "unconditional"  CFU  "deflection immediate response"  CD(i,l)  183 Session Progress encapsulated ACM generic notification indicator "call is diverting"  NOTIFY: Notification indicator "call is diverting"								
Comments	ISDN	SUT	SIP-I						
	SETUP -	<b>→</b>	INVITE(IAM)						
	NOTIFY <b>←</b>	+	183 Session Progress(ACM)						
	ALERTING ←	+	180 Ringing(CPG)						
	CONNECT								
		<b>→</b>	ACK						
	DISCONNECT →	<b>→</b>	BYE(REL)						
	RELEASE ←	+	200 OK BYE(RLC)						
	RELEASE COMPLETE →								

TP712002	SIP reference: RFC 326	1 [4]		ISUP reference: Q.1912.5 [1], clause 2.5.2.1.1/Q.732 [i.4]				
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV							
SIP selection								
criteria								
ISUP selection								
criteria								
Test purpose  SIP parameter values	diversion information and the Applicable redirection reason in "busy" "no reply" "deflection during alerting" "deflection immediate response"	essfully expression of the call of the cal	stablished in the optic ication in on numbe liversion FB(u,e) FNR O(a) O(i,e) backward	onal bodicater, if dinforr	packward call indicators. The or set to "call is diverting", the call iversion occurs.  mation:			
ISDN parameter	call diversion information, redire	ction num	ber					
values Comments	ISDN		SUT		SIP-I			
Comments	SETUP	<b>→</b>	301	<b>→</b>	INVITE(IAM)			
	ALERTING	-		<del></del>	180 Ringing(ACM)			
	ALLICINO	+ • +		÷	183 Session Progress(CPG)			
	CONNECT	+		÷	200 OK INVITE(ANM)			
	001111201			<u>`</u>	ACK			
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)			
	RELEASE	+		+	200 OK BYE(RLC)			
	RELEASE COMPLETE	<b>→</b>						

TP712003	SIP reference: RFC 3	261 [4]		ISUP reference: Q.1912.5 [1]						
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV									
SIP selection										
criteria										
ISUP selection criteria										
Test purpose	Redirection number - preser	ntation all	owed - acc	ordir	ng to the notification					
	subscription option									
	To verify that the originating e	xchange n	nakes the re	edirec	ction number available to the					
					ription option of the call diversion					
	information is coded "010 pres									
	The redirection number restrict									
SIP parameter					ation indicator "call is diverting"					
values	200 OK INVITE: encapsulated	I ANM red	irection num	nber i	estriction "presentation allowed"					
ISDN parameter values	CONNECT: redirection number	er								
Comments	ISDN		SUT		SIP-I					
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)					
	NOTIFY	+		+	183 Session Progress(ACM)					
	ALERTING	+		+	180 Ringing(CPG)					
	CONNECT	+		+	200 OK INVITE(ANM)					
				<b>→</b>	ACK					
			I							
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)					
	RELEASE	+		+	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>→</b>								

TP712004	SIP reference: RFC 3261	[4]	claus	ISUP reference: Q.1912.5 [1], clause 2.4.2; table 2-1/Q.732 [i.4]					
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Redirection number - presentation restricted - according to the notification subscription option  To verify that the originating exchange does not make the redirection number available to the calling access signalling system, if the notification subscription option of the call diversion information is coded "001 presentation not allowed", "011 presentation allowed without redirection number" or "000 unknown".								
SIP parameter	183 Session Progress encapsulate	ed ACM	call diversion i	nformation notification					
values	subscription option "presentation								
ISDN parameter values	NOTIFY: notification indicator "cal ALERTING: no redirection numbe CONNECT: no redirection number	1	ted"						
Comments	ISDN		SUT	SIP-I					
	SETUP	<b>→</b>	→	INVITE(IAM)					
	NOTIFY	+	+	183 Session Progress(ACM)					
	ALERTING	+	+	180 Ringing(CPG)					
	CONNECT	<del>-</del>	+	200 OK INVITE(ANM)					
			<b>→</b>	ACK					
	DISCONNECT	<b>→</b>	<b>→</b>	BYE(REL)					
	RELEASE	<del>-</del>	+	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>→</b>							

TP712005	SIP reference: RFC 3261	[4]	ISUP reference: Q.1912.5 [1], clause 2.4.2; table 2-1/Q.732 [i.4]						
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Redirection number - presentation restricted - according to redirection number restriction parameter  To verify that the originating exchange does not make the redirection number available to the calling access signalling system, if the redirection number restriction parameter indicates "01 Presentation restricted".  The notification subscription option of the call diversion information is coded "010 Presentation allowed with redirection number".								
SIP parameter values	183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed with redirection number" 200 OK INVITE: encapsulated ANM redirection number restriction "presentation restricted"								
ISDN parameter values	CONNECT: no redirection number	r							
Comments	ISDN	SU	ΙΤ	SIP-I					
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)					
	NOTIFY	+	+	183 Session Progress(ACM)					
	ALERTING	+	+	180 Ringing(CPG)					
	CONNECT ← 200 OK INVITE(ANM)								
			→	ACK					
	DISCONNECT	<b>→</b>	<b>→</b>	BYE(REL)					
	RELEASE	+	+	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>→</b>							

TP712006	SIP reference: RFC 3261 [4]		claus	ISUP reference: Q.1912.5 [1], se 2.4.2; table 2-1/Q.732 [i.4]					
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV	-							
SIP selection									
criteria									
ISUP selection criteria									
SIP parameter values ISDN parameter values	Redirection number - presentation restricted - no redirection number restriction parameter received  To verify that the originating exchange does not make the redirection number available to the calling access signalling system, if no redirection number restriction parameter is received.  The notification subscription option of the call diversion information is coded "010 Presentation allowed with redirection number".  183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed with redirection number"  200 OK INVITE: encapsulated ANM without redirection number restriction parameter								
Comments	ISDN	SUT		SIP-I					
	SETUP →		<b>→</b>	INVITE(IAM)					
	NOTIFY		<del>-</del>	183 Session Progress(ACM)					
	ALERTING +		+	180 Ringing(CPG)					
	CONNECT		+	200 OK INVITE(ANM)					
		<u> </u>	<b>→</b>	ACK					
	DISCONNECT →		<b>→</b>	BYE(REL)					
	RELEASE +		+	200 OK BYE(RLC)					
	RELEASE COMPLETE →								

TP712007	SIP reference: RFC 326	61 [4]		ISUP reference: Q.1912.5 [1], clause 2.4.2/Q.732 [i.4]						
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV									
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	To verify that the originating exc	Multiple diversions - redirection number not send by the last diversion  To verify that the originating exchange does not make any redirection number available to the calling access signalling system, if the last diverting exchange does not send one (see note)								
SIP parameter	183 Session Progress encapsul	ated ACM	call diversion	n information notification						
values	subscription option "presentation allowed with redirection number", redirection number 183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed with redirection number", no redirection number 200 OK INVITE: encapsulated ANM redirection number restriction "presentation allowed"									
ISDN parameter values	ALERTING: no redirection number CONNECT: no redirection number 1									
Comments	ISDN		SUT	SIP-I						
	SETUP	<b>→</b>	<b>→</b>	INVITE(IAM)						
	NOTIFY	+	+	183 Session Progress(ACM)						
			+	183 Session Progress(CPG)						
	ALERTING	+	+	180 Ringing(CPG)						
	CONNECT ← 200 OK INVITE(ANM)  → ACK									
	DISCONNECT	<b>→</b>	-	BYE(REL)						
	RELEASE	+	<del>-</del>							
NOTE: The first	RELEASE COMPLETE	<b>→</b>		United the ite and extended. The						

NOTE: The first diverting exchange sends the **redirection number** and allows for its presentation. The second (last) diversion allows for the presentation of the **redirection number**, but does not send it, i.e. only **call diversion information** is present in the message and the redirection number is missing. The **redirection number restriction** parameter is also received as "presentation allowed".

TP712008	SIP reference: RFC 3261	[4]		ISUP reference: Q.1912.5 [1], clause 2.4.2/Q.732 [i.4]					
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV								
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Multiple diversions - redirection number - presentation according to the most restrictive notification subscription option  To verify that the originating exchange handles the presentation of the redirection number according to the contents of the most restrictive notification subscription option of the call diversion information, if the forwarded-to user allows presentation of the number ("presentation allowed" in the redirection number restriction parameter) (see note).								
SIP parameter	183 Session Progress encapsulate								
values	subscription option "presentation allowed with redirection number", redirection number 183 Session Progress encapsulated ACM call diversion information notification subscription option "presentation allowed without redirection number", redirection number 200 OK INVITE: encapsulated ANM redirection number restriction "presentation allowed"								
ISDN parameter	ALERTING: no redirection number								
values	CONNECT: no redirection number								
Comments	ISDN		SUT		SIP-I				
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)				
	NOTIFY	<del>-</del>		+	183 Session Progress(ACM)				
				+	183 Session Progress(CPG)				
	ALERTING	<del>-</del>		+	180 Ringing(CPG)				
	CONNECT	<del>(</del>		+	200 OK INVITE(ANM)				
				<b>→</b>	ACK				
	DISCONNECT	<b>→</b>		<b>→</b>	BYE(REL)				
	RELEASE	<del>(</del>		+	200 OK BYE(RLC)				
	RELEASE COMPLETE	<b>→</b>							
forwardin	TREED, RECORDING DE LE								

TP712009	SIP re	eference: RF	C 32	261 [4]	ISUP reference: Q.1912.5 [1], clause 2.5.2.5.1.1/Q.732 [i.4]			
TSS reference	ISDN-(ISUP)-	-SIP/SS/CDI\	/					
SIP selection criteria								
ISUP selection criteria								
Test purpose	To verify that	the IUT acce	pts a		fully establish a di			
SIP parameter	183 Session	Progress: end	caps	ulated ACM gen	eric notification "ca	all i	is diverted", redirection	
values	information, r	edirection nu	mbe	r				
ISDN parameter values								
Comments	ISDN 2	SUT		SIP-I 1		•	SIP-I 3	
			<b>←</b>	INVITE(IAM)				
		CDIV						
			<b>→</b>	183 Session Pr	ogress(ACM)			
					-	<b>→</b>	NVITE(IAM)	
					•	L	180 Ringing(ACM)	
			<b>→</b>	180 Ringing(AC	CM)			
					•	<b>E</b> 2	200 OK INVITE(ANM)	
			<b>→</b>	200 OK INVITE	(ANM)	<b>&gt;</b> /	ACK	
			+	ACK				
			+	BYE(REL)		<b>→</b> [	BYE(REL)	
			<b>→</b>	200 OK BYE(R	LC •		200 OK BYE(RLC	

TP712010	SIP reference: RFC 326	[4]	cla	ISUP reference: Q.1912.5 [1], clause 2.5.2.5.1.1/Q.732 [i.4]					
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Setting of redirection number r (pres. allowed) To verify that the IUT includes the CPG, ANM or CON set to "prese	e redirec	tion number r	restriction indicator in the ACM,					
SIP parameter values	200 OK INVITE: encapsulated ANM redirection number restriction "presentation allowed"								
ISDN parameter values									
Comments	ISDN		SUT	SIP-I					
	SETUP	+	<b>+</b>	INVITE(IAM)					
	ALERTING	<b>→</b>	→	180 Ringing(ACM)					
	CONNECT	<b>→</b>	→	200 OK INVITE(ANM)					
		← ACK							
	DISCONNECT	+	+	BYE(REL)					
	RELEASE	<b>→</b>	<b>→</b>	200 OK BYE(RLC)					
	RELEASE COMPLETE	<b>←</b>							

TP712011	SIP reference: RFC 3	261 [4]		ISUP reference: Q.1912.5 [1],
			cla	use 2.5.2.5.1.1/Q.732 [i.4]
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV			
SIP selection criteria				
ISUP selection criteria				
Test purpose	Setting the redirection num (pres. restricted) To verify that the IUT includes CPG, ANM or CON set to "pre	the <b>redirectio</b>	n number r	estriction indicator in the ACM,
SIP parameter values	200 OK INVITE: encapsulated	ANM redirection	on number r	restriction "presentation restricted"
ISDN parameter values				
Comments	ISDN	SU	JT	SIP-I
	SETUP	+	+	INVITE(IAM)
	ALERTING	<b>→</b>	→	180 Ringing(ACM)
	CONNECT	<b>→</b>	<b>→</b>	200 OK INVITE(ANM)
			+	ACK
	DISCONNECT	<del>-</del>	<b>←</b>	BYE(REL)
	RELEASE	→	→	200 OK BYE(RLC)
	RELEASE COMPLETE	<b>←</b>		

TP712012	SIP reference: RFC 3261 [4]				ISUP reference: Q.1912.5 [1], clause 2.5.2.5.1.2 b) 2)/Q.732 [i.4]				
TSS reference	ISDN-(ISUP)	-SIP/SS/CDI\	<b>/</b>						
SIP selection									
criteria									
ISUP selection criteria									
Test purpose	Verify that th number according	e IUT sets the ording to the	ado 'serv	rated by the diverting exchangeress presentation restricted induced user releases his/her number	icato er to	the diverted-to user"			
SIP parameter	INVITE SIP-I	3:encapsulat	ted I/	AM original called number prese	entat	ion allowed			
values		•							
ISDN parameter									
values									
Comments	ISDN 2	SUT		SIP-I 1		SIP-I 3			
			<b>←</b>	INVITE(IAM)					
		CDIV							
			<b>→</b>	183 Session Progress(ACM)					
					<b>→</b>	INVITE(IAM)			
					+	180 Ringing(ACM)			
			<b>→</b>	180 Ringing(ACM)					
					+	200 OK INVITE(ANM)			
			→	200 OK INVITE(ANM)		ACK			
				ACK					
			- 1			I			
			<b>←</b>	BYE(REL)	<b>→</b>	BYE(REL)			
			<b>→</b>	200 OK BYE(RLC	+	200 OK BYE(RLC			

TP712013	SIP r	eference: RF	C 32	261 [4] I		reference: 912.5 [1]
TSS reference	ISDN-(ISUP)	-SIP/SS/CDIV	/			
SIP selection criteria						
ISUP selection criteria						
Test purpose	Verify that th number according	e IUT sets the ording to the	e ado 'serv	ed by the diverting exchange dress presentation restricted in ed user releases his/her numb redirection information shall	dicato er to	the diverted-to user"
SIP parameter values	INVITE SIP-	l 3: redirectino	g nun	nber, redirection information		
ISDN parameter values						
Comments	ISDN 2	SUT		SIP-I 1		SIP-I 3
			+	INVITE(IAM)		
		CDIV				
			<b>→</b>	183 Session Progress(ACM)		
					<b>→</b>	INVITE(IAM)
					+	180 Ringing(ACM)
			<b>→</b>	180 Ringing(ACM)		
					+	200 OK INVITE(ANM)
			<b>→</b>	200 OK INVITE(ANM)	<b>→</b>	ACK
			+	ACK		
			+	BYE(REL)	<b>→</b>	BYE(REL)
			→	200 OK BYE(RLC	+	200 OK BYE(RLC

TP712014	SIP re	eference: RF	C 32		Q.19	reference: 912.5 [1], 1.2 b) 5)/Q.732 [i.4]
TSS reference	ISDN-(ISUP)-	SIP/SS/CDI	/			
SIP selection criteria						
ISUP selection criteria						
Test purpose	To verify that indicator rece - not require - preferred	the IUT can ived in the formal the war all the way sall t	succ <b>orwa</b> y" sh shall	dicator in the diverting excha essfully divert a call and that IS rd call indicators with the valuall be changed to "ISDN user pate left unchanged; be left unchanged.	SDN i ie "IS	user part preference SDN user part:
SIP parameter				IAM forward call indicator ISDN	l use	er part required all the
values	way	•				
ISDN parameter values						
Comments	ISDN 2	SUT		SIP-I 1		SIP-I 3
			+	INVITE(IAM)		
		CDIV				
			<b>→</b>	183 Session Progress(ACM)		
					<b>→</b>	INVITE(IAM)
					+	180 Ringing(ACM)
			<b>→</b>	180 Ringing(ACM)		
					+	200 OK INVITE(ANM)
			→	200 OK INVITE(ANM)	→	ACK
			<b>←</b>	ACK		
			<b>←</b>	BYE(REL)	<b>→</b>	BYE(REL)
			<b>→</b>	200 OK BYE(RLC	+	200 OK BYE(RLC

TP712015	SIP reference: RFC 326	1 [4]	clause	ISUP reference: Q.1912.5 [1], 2.5.2.5.1.2 c) ii); iii)/Q.732 [i.4]
TSS reference	ISDN-(ISUP)-SIP/SS/CDIV			, ,, ,
SIP selection criteria				
ISUP selection criteria				
Test purpose	Call diversion may occur in the To verify that the IUT includes ar "call diversion may occur" in the	optional l	oackward ca	
SIP parameter values	180 Ringing: encapsulated ACM backward call indicator "call dive			ator "subscriber free" optional
ISDN parameter values	ALERTING: no mapping of option			or value
Comments	ISDN	5	SUT	SIP-I
	SETUP	<b>→</b>	→	INVITE(IAM)
	ALERTING	+	+	180 Ringing(ACM)
	CONNECT	+	+	200 OK INVITE(ANM)
			→	ACK
	DISCONNECT	<b>→</b>	→	BYE(REL)
	RELEASE	+	+	200 OK BYE(RLC)
	RELEASE COMPLETE	+		

### A.1.1.2.13 Call HOLD (HOLD)

TP713001	SIP reference	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 2.5.2.1.1.1; 2.5.2.1.1.2/Q.733 [i.6]		
TSS reference	ISDN-(ISUP)-SIP/SS/I	HOLD					
SIP selection							
criteria							
ISUP selection criteria							
Test purpose	Call hold after answer To verify that a call cathat notifications are supprogress.	in be placed or	n hold and	l can be retrie	ved again by the local user and vent indicator set to		
SIP parameter values							
ISDN parameter values							
Comments	ISDN		SL	JT	SIP		
	SETUP	<b>→</b>		→	INVITE(IAM)		
	ALERTING	+		+	180 Ringing(ACM)		
	CONN	+		+	200 OK INVITE(ANM)		
				→	ACK		
			Commur	nication			
	HOLD	<b>→</b>		→	INFO(CPG hold)		
				<b>←</b>	200 OK INFO		
	RETRIVE	<b>→</b>		<b>→</b>	INFO(CPG retrieve)		
				+	200 OK INFO		
			Commur				
	DISC	<b>→</b>		<b>→</b>	BYE(REL)		
	REL	+		+	200 OK BYE		
	REL_COM	→					

TP713002	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 2.5.2.1.1.1; 2.5.2.1.1.2/Q.733 [i.6]		
TSS reference	ISDN-(ISUP)-SIP/SS/F	HOLD				
SIP selection criteria						
ISUP selection criteria						
Test purpose	Call hold after answe To verify that a call car and that notifications a	n be placed on	hold and	can be		ed again by the remote user
SIP parameter values						
ISDN parameter values						
Comments	ISDN		SU	T		SIP
	SETUP	<b>→</b>			<b>→</b>	INVITE(IAM)
	ALERTING	+			+	180 Ringing(ACM)
	CONN	+			+	200 OK INVITE(ANM)
					<b>→</b>	ACK
			Commun	ication		
	HOLD	<b>←</b>			+	INFO(CPG hold)
					→	200 OK INFO
	RETRIVE	<b>←</b>			+	INFO(CPG retrieve)
					→	200 OK INFO
			Commun	ication		
	DISC	→			→	BYE(REL)
	REL	+			+	200 OK BYE
	REL_COM	→				

TP713003	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clauses 2.2.1; 2.5.2.1.1.1; 2.5.2.1.1.2/Q.733 [i.6]				
TSS reference	ISDN-(ISUP)-SIP/SS/I	HOLD						
SIP selection criteria								
ISUP selection criteria	PICS 8/1							
Test purpose	To verify that an outgo	Call hold after alerting, requested by the local user To verify that an outgoing call can be placed on HOLD after alerting has commenced and can be retrieved afterwards by the local user and that notifications are sent with CPG messages						
SIP parameter								
values								
ISDN parameter values								
Comments	ISDN		SU	IT	SIP			
	SETUP	<b>→</b>		-	► INVITE(IAM)			
	ALERTING	+		•	180 Ringing(ACM)			
	HOLD	<b>→</b>		-	NFO(CPG hold)			
				•	200 OK INFO			
	RETRIVE	<b>→</b>		-	► INFO(CPG retrieve)			
				•	200 OK INFO			
	CONN	+		•	200 OK INVITE(ANM)			
				-	► ACK			
			Commur	nication				
	DISC	<b>→</b>		<b>-</b>	▶ BYE(REL)			
	REL	+		•	200 OK BYE			
	REL_COM	<b>→</b>	-					

TP713004	SIP reference: RFC 3261 [4]			clauses	ISUP reference: Q.1912.5 [1], 2.2.1; 2.5.2.5.1/Q.733 [i.6]
TSS reference	ISDN-(ISUP)-SIP/SS/	HOLD			
SIP selection criteria					
ISUP selection criteria	PICS 8/1				
Test purpose	Call hold after alerting To verify that an incorrupt the remote user.				an be retrieved afterwards by
SIP parameter					
values					
ISDN parameter values					
Comments	ISDN		SU	Т	SIP
	SETUP	→		→	INVITE(IAM)
	ALERTING	+		+	180 Ringing(ACM)
	HOLD	+		+	INFO(CPG hold)
				<b>→</b>	200 OK INFO
	RETRIVE	+		+	INFO(CPG retrieve)
				<b>→</b>	200 OK INFO
	CONN	+		+	200 OK INVITE(ANM)
				<b>→</b>	ACK
			Commun	ication	
	DISC	<b>→</b>		→	BYE(REL)
	REL	+		+	200 OK BYE
	REL_COM	<b>→</b>			

TP713005	SIP reference:			(	-	SUP reference: Q.1912.5 [1], se 2.3/Q.764 [i.12]
TSS reference	ISDN-(ISUP)-SIP/SS/H	OLD				
SIP selection criteria						
ISUP selection criteria						
Test purpose	Call hold after answer To verify that a call in the hold service.					ved user ser who activated the Call
SIP parameter values						
ISDN parameter values						
Comments	ISDN		SU	Т		SIP
	SETUP	→		-	<b>→</b>	INVITE(IAM)
	ALERTING	+		•	<del>(</del>	180 Ringing(ACM)
	CONN	+		•	<del>(</del>	200 OK INVITE(ANM)
				-	<b>→</b>	ACK
		(	Commun	ication		
	HOLD	<b>→</b>		-	<b>→</b>	INFO(CPG hold)
				•	<del>(</del>	200 OK INFO
	DISC	<b>→</b>			<del>)</del>	BYE(REL)
	REL	+		•	<u> </u>	200 OK BYE
	REL_COM	→				

TP713006	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 2.3/Q.764 [i.12]		
TSS reference	ISDN-(ISUP)-SIP/SS/	HOLD				
SIP selection criteria						
ISUP selection criteria						
Test purpose	Call hold after answer To verify that a call in Call hold service.				ved user user who did not activate the	
SIP parameter values						
ISDN parameter values						
Comments	ISDN		SUT		SIP	
	SETUP	<b>→</b>		→	INVITE(IAM)	
	ALERTING	+		+	180 Ringing(ACM)	
	CONN	+		+	200 OK INVITE(ANM)	
				→	ACK	
		С	ommunication	1		
	HOLD	<b>→</b>		→	INFO(CPG hold)	
				+	200 OK INFO	
	DISC	+		+	BYE(REL)	
	REL	<b>→</b>		+	200 OK BYE	
	REL_COM	+				

TP713007	SIP reference: RFC 3261 [4]			ISUP reference: Q.1912.5 [1], clause 2.3/Q.764 [i.12]		
TSS reference	ISDN-(ISUP)-SIP/SS/HOL	_D				
SIP selection						
criteria						
ISUP selection criteria						
Test purpose	Call hold after alerting, I To verify that a held call c without retrieving the call.	an be relea	•		ved user ivated the Call hold service	
SIP parameter values						
ISDN parameter values						
Comments	ISDN		SUT		SIP	
	SETUP	<b>+</b>		+	INVITE(IAM)	
	ALERTING	→		<b>→</b>	180 Ringing(ACM)	
	HOLD	→		<b>→</b>	INFO(CPG hold)	
				+	200 OK INFO	
	DISC	<b>→</b>		<b>→</b>	CANCEL/BYE	
	RELEASE	+		+	200 OK CANCEL/BYE	
	REL_COMP	<b>→</b>		+	487 Request Terminated	
				<b>→</b>	ACK	

TP713008	SIP reference: I	RFC 3261 [4]		ISUP reference: Q.1912.5 [1], clause 2.3/Q.764 [i.12]								
TSS reference	ISDN-(ISUP)-SIP/SS/HO	OLD										
SIP selection												
criteria												
ISUP selection												
criteria												
Test purpose		Call hold after answer, release of the call by the non-served user										
	To verify that a call in th	e held state ca	ın be released	d by the ι	ser who did not activate the							
	Call hold service.											
SIP parameter												
values												
ISDN parameter values												
Comments	ISDN		SUT		SIP							
	SETUP	+		+	INVITE(IAM)							
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)							
	HOLD	+		+	INFO(CPG hold)							
				<b>→</b>	200 OK INFO							
	DISC	+		+	CANCEL							
	RELEASE	<b>→</b>		<b>→</b>	200 OK CANCEL/BYE							
	REL_COMP	+		<b>→</b>	487 Request Terminated							
				+	ACK							

### A.1.1.2.14 Call Waiting (CW)

TP714001	SIP reference: I	RFC 3261 [4]			SUP reference: Q.1912.5 [1], e 1.5.2.1.1/Q.733 [i.6]					
TSS reference	ISDN-(ISUP)-SIP/SS/C\	N								
SIP selection criteria										
ISUP selection criteria										
Test purpose	Call waiting indication To verify that a call can call.		ly established if	f the <b>AC</b>	<b>M</b> indicates that it is a waiting					
SIP parameter values	180 Ringing: encapsular a waiting call"	180 Ringing: encapsulated ACM contains the Generic notification parameter value "call is								
ISDN parameter values										
Comments	ISDN		SUT		SIP					
	SETUP	<b>→</b>		<b>→</b>	INVITE(IAM)					
	ALERTING	+		+	180 Ringing(ACM)					
	CONN	+		+	200 OK INVITE(ANM)					
				<b>→</b>	ACK					
	Communication									
	DISC	<b>→</b>		<b>→</b>	BYE(REL)					
	REL	+		+	200 OK BYE					
	REL_COM	<b>→</b>								

TP714002	SIP reference: RF	C 3261 [4]		c	ISUP reference: Q.1912.5 [1], clause 1.5.2.1.1/Q.733 [i.6]				
TSS reference	ISDN-(ISUP)-SIP/SS/CW								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Call waiting indication in To verify that a call can be call.		Illy establ	ished if the	e CPG indicates that it is a waiting				
SIP parameter values	180 Ringing: encapsulated ACM the called party status is set to "no indication" 183 Session Progress: encapsulated CPG Alerting contains the Generic notification parameter value "call is a waiting call"								
ISDN parameter values									
Comments	ISDN		SUT	-	SIP				
	SETUP	<b>→</b>		→	INVITE(IAM)				
				+	183 Session Progress(ACM)				
	ALERTING	+		+	180 Ringing(CPG)				
	CONN	+		+	200 OK INVITE(ANM)				
				→	ACK				
		С	ation						
	DISC	→		→	BYE(REL)				
	REL	+		<b>←</b>	200 OK BYE				
	REL_COM	→							

TP714003	SIP reference: RFC	3261 [4]		ISUP reference: Q.1912.5 [1], clause 1.5.2.5.1/Q.733 [i.6]						
TSS reference	ISDN-(ISUP)-SIP/SS/CW									
SIP selection criteria										
ISUP selection criteria										
Test purpose	Call waiting indication in ACM or CPG To verify that a call can be successfully established if the user has subscribed to the call waiting service (with notification) and if he is currently busy, but answers the waiting call. The indication shall be sent either in an ACM or a CPG.									
SIP parameter	180 Ringing: encapsulated A	ACM contain	ns the Generic	notific	ation parameter value "call is					
values	a waiting call"				·					
ISDN parameter values										
Comments	ISDN		SUT		SIP					
	SETUP	+		+	INVITE(IAM)					
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)					
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)					
				+	ACK					
	DISC	+		+	BYE(REL)					
	REL				200 OK BYE					
	REL_COM	+	•							

TP714004	SIP reference: RF	C 326	1 [4]		ISUP reference: Q.1912.5 [1], clause 1.5.2.5.2/Q.733 [i.6]				
TSS reference	ISDN-(ISUP)-SIP/SS/CW								
SIP selection criteria									
ISUP selection criteria									
Test purpose	Call waiting rejected To verify that the IUT sends a REL with cause #21 (call rejected) if a busy user rejects the waiting call.								
SIP parameter values	480 Temporarily unavailable: encapsulated REL cause 21								
ISDN parameter values	RELEASE COMPLETE: ca	ause 2	1						
Comments	ISDN		SUT		SIP				
	SETUP	+		+	INVITE(IAM)				
	ALERTING	<b>→</b>		<b>→</b>	180 Ringing(ACM)				
	CONN	<b>→</b>		<b>→</b>	200 OK INVITE(ANM)				
				<b>←</b>	ACK				
		C	ommunicati	on					
	SETUP	<b>←</b>		+	INVITE(IAM)				
	ALERTING	<b>→</b>		→	180 Ringing(ACM waiting call)				
	RELEASE COMPLETE	<b>→</b>		→	480 Temporarily unavailable(REL#21)				
				<b>←</b>	ACK				
	DISC	+		+	BYE(REL)				
	REL				200 OK BYE				
	REL_COM	<b>←</b>							

### A.1.1.2.15 Three Party Service (3PTY)

TP715001	SIP refe	erence	e: RFC 32	61 [·		ISUP reference: Q.1912.5 [1], clauses 2.4; 2.2.1/Q.734.2 [i.8]						
TSS reference	ISDN-(ISUP)-S	IP/SS/	3PTY		•							
SIP selection												
criteria												
ISUP selection												
criteria												
Test purpose	To verify that the successfully join remote parties. The IUT should	Served user initiates 3PTY  To verify that the IUT, where the served user with two active calls is located, can successfully join these calls to form a three-way conversation, and notify the implied remote parties accordingly.  The IUT should send CPG messages with the generic notification indicator set to "conference established" to both implied parties. The event indicator in the CPG should be set to "progress"										
SIP parameter	be set to progr											
values												
ISDN parameter values												
Comments	ISDN	SUT			SIP-I 1		SIP-I 2					
	SETUP	<b>→</b>		1	INVITE(IAM)							
	ALERTING	+		+	180 Ringing(ACM)							
	CONN	<b>←</b> 200 OI		200 OK INVITE(ANM)								
	Communication											
	HOLD	<b>→</b>		<b>→</b>	INVITE(CPG hold)							
				+	200 OK INVITE							
				<b>→</b>	ACK							
	SETUP	<b>→</b>				→	INVITE(IAM)					
	ALERTING	+				+	180 Ringing(ACM)					
	CONN	+				+	200 OK INVITE(ANM)					
	FAC(est3pty)	<b>→</b>		<b>→</b>	INFO(CPG conf est)							
				+	200 OK INFO							
						<b>→</b>	INFO(CPG conf est)					
					<u></u>	<b>←</b>	200 OK INFO					
			<del></del>		PTY communication		T					
	DISC	+		<del>(</del>	BYE(REL)							
	RELEASE	<b>→</b>		<b>→</b>	200 OK BYE							
	REL_COM	+				<b>→</b>	INFO(CPG conf disc)					
						<b>←</b>	200 OK INFO					
	DISC	<b>→</b>				<b>→</b>	BYE(REL)					
	RELEASE	+				+	200 OK BYE					
	REL_COM	<b>→</b>										

TP715002	SIP refe	rence	e: RFC 3261 [			reference:				
					alausa 2		912.5 [1],			
TSS reference	clause 2.5.2.1.1.3 a)/Q.734.2 [i.8]   ISDN-(ISUP)-SIP/SS/3PTY									
SIP selection	13014-(1307)-31	F/33/	3511							
criteria										
ISUP selection										
criteria										
Test purpose	Served user cr	eates	a private coi	nmunicati	on with a rem	ote u	iser			
	To verify that the IUT (controlling the conference) on a 3PTY call can successfully create									
	private communication with one of the remote users. The appropriate notification									
	(depending on A-B active-held or A-C active-idle connection) is sent in <b>CPG</b> messages to									
CID managed an	the two users.									
SIP parameter values										
ISDN parameter										
values										
Comments	ISDN SUT SIP-I 1						SIP-I 2			
	SETUP	<b>→</b>	→	INVITE(I	AM)		<u> </u>			
	ALERTING	+	+		ing(ACM)					
	CONN	+	+		NVITE(ANM)					
			Communicat		, ,					
	HOLD	<b>→</b>	→	INVITE(0	CPG hold)					
			+	200 OK I	NVITE					
			<b>→</b>	ACK						
	SETUP	<b>→</b>				<b>→</b>	INVITE(IAM)			
	ALERTING	+				+	180 Ringing(ACM)			
	CONN	+				+	200 OK INVITE(ANM)			
	FAC(est3pty)	<b>→</b>	→		PG conf est)					
			<b>←</b>	200 OK I	NFO					
						<b>→</b>	INFO(CPG conf est)			
				<u></u>		<b>←</b>	200 OK INFO			
	EAG( 10 ( )				nunication	_				
	FAC(end3pty)	<b>→</b>	<b>→</b>		PG conf disc)	-				
	FAC(ret res)	<b>←</b>	+	200 OK I	NFO	+	INFO(ODO(-/' )			
		+				<b>→</b>	INFO(CPG conf disc) 200 OK INFO			
		-	+	Com	munication IS					
	DISC	+	<b>+</b>	BYE(REI		   	JIF-1 Z			
	RELEASE	→ →		200 OK						
	REL_COM	<del>-</del>		200 OK I	J   L					
	DISC	→	+			<b>→</b>	BYE(REL)			
	RELEASE	<del>-</del>				<del>-</del>	200 OK BYE			
	REL_COM	→ ·				+	200 OR DIL			

TP715003	SIP refe	: RFC 3261 [	ISUP reference: Q.1912.5 [1], clause 2.5.2.1.1.3 b)/Q.734.2 [i.8]								
TSS reference	ISDN-(ISUP)-SIP/SS/3PTY										
SIP selection											
criteria											
SUP selection											
riteria											
Test purpose	To verify that the disconnect one messages. The IUT should notification income.	Served user disconnects one remote user and retains the other To verify that the IUT (controlling the conference) on a 3PTY call can successfully disconnect one remote user and retain and notify the other user appropriately using CPG messages. The IUT should send to the appropriate remote users CPG messages with a generic notification indicator (depending on A-B active-held or A-C active-idle connection). The event indicator in the CPG should be set to "progress" (see note).									
SIP parameter											
values											
SDN parameter											
values											
Comments	ISDN		SUT	SIP-I 1			SIP-I 2				
	SETUP	<b>→</b>	→	INVITE(IAM)							
	ALERTING	+	<b>←</b>	180 Ring	jing(ACM)						
	CONN	+	<b>←</b>	200 OK I	NVITE(ANM)						
	HOLD	<b>→</b>	→	INVITE(0	CPG hold)						
			<del>(</del>	200 OK I	NVITE						
			→	ACK							
	SETUP	<b>→</b>				<b>→</b>	INVITE(IAM)				
	ALERTING	+				+	180 Ringing(ACM)				
	CONN	+				+	200 OK INVITE(ANM)				
	FAC(est3pty)	<b>→</b>	→	INFO(CF	PG conf est)						
	\ 1 2/		+	200 OK I							
						<b>→</b>	INFO(CPG conf est)				
						+	200 OK INFO				
			3	PTY com	munication		•				
	DISC	<b>→</b>				<b>→</b>	BYE(REL)				
	RELEASE	+				+	200 OK BYE				
	REL_COM	<b>→</b>	<b>→</b>	INFO(CF	PG conf disc)						
	_		<del>(</del>	200 OK I							
			<b>→</b>	INFO(CF	PG hold)						
			<b>←</b>	200 OK I							
	DISC	+	<del>(</del>	BYE(REI							
	RELEASE	<b>→</b>	<b>→</b>	200 OK I	,						
	REL_COM	+									
NOTE: The "ren "confere		tion sho " notific	ould be sent i cation in a se	n a CPG to parate CPC	o the remaining <b>3.</b>	rem	ote user, followed by the				

TP715004	SIP refe	rence	: RFC 326	1 [4	ij	1		reference: 912.5 [1],	
						clause		.1.1.3/Q.734.2 [i.8]	
TSS reference	ISDN-(ISUP)-SI	IP/SS/3	3PTY		I				
SIP selection	, ,								
criteria									
ISUP selection criteria									
Test purpose	Served user disconnects both remote users and terminates the call To verify that the IUT (controlling the conference) can send the appropriate notification to the two remote users when disconnecting both remote users on the 3PTY call. The IUT should send to the appropriate remote users a CPG with a generic notification indicator (depending on A-B active-held or A-C active-idle connection). The event indicator in the CPG is set to "progress".								
SIP parameter values			•						
ISDN parameter values									
Comments	ISDN	SUT			SIP-I 1			SIP-I 2	
	SETUP(CRx)	<b>→</b>	-	<b>&gt;</b>	INVITE(IA	M)			
	ALERTING	+	•	-	180 Ringir				
	CONN	+	•	-	200 OK IN	IVITE(ANM)			
	HOLD	<b>→</b>		<b>&gt;</b>	INVITE(CI				
			•	-	200 OK IN	IVITE			
			]	>	ACK				
	SETUP(CRy)	<b>→</b>					<b>→</b>	INVITE(IAM)	
	ALERTING	+					+	180 Ringing(ACM)	
	CONN	<b>←</b>					<b>←</b>	200 OK INVITE(ANM)	
	FAC(est3pty)	<b>→</b>	-		INFO(CPC				
			•	_	200 OK IN	IFO			
							<b>→</b>	INFO(CPG conf est)	
							<b>←</b>	200 OK INFO	
				3 I	PTY comm				
	DISC(CRx)	<b>→</b>		<b>&gt;</b>	BYE(REL)				
	RELEASE	+	(	-	200 OK B	YE			
	REL_COM	<b>→</b>					<b>→</b>	INFO(CPG conf disc)	
							+	200 OK INFO	
	DISC(CRy)	<b>→</b>					<b>→</b>	BYE(REL)	
	RELEASÉ	+					+	200 OK BYE	
	REL_COM	<b>→</b>							

TP715005	SIP refe	: RFC 326	1 [4	4]	ISUP reference: Q.1912.5 [1], clause 2.2.1/Q.734.2 [i.8]						
TSS reference	ISDN-(ISUP)-SIP/SS/3PTY										
SIP selection											
criteria											
ISUP selection											
criteria											
Test purpose	Remote user disconnects 3PTY call To verify that the IUT (controlling the conference) can successfully continue the 3PTY call after receiving disconnection by one of the remote users, and send the appropriate notification to the remaining party. The IUT should send to the other remote user CPG with a generic notification indicator (depending on A-B active-held or A-C active-idle connection). The event indicator in the CPG is set to "progress" (see note).										
SIP parameter											
values											
ISDN parameter											
values	10011				lois i		1	Torn v. a			
Comments	ISDN		SUT		SIP-I 1			SIP-I 2			
	SETUP	<b>→</b>		<u> </u>	INVITE(IAN						
	ALERTING	+		<u> </u>	180 Ringing						
	CONN	<b>←</b>	<u> </u>	<u> </u>	200 OK IN\	<u>/ITE(ANM)</u>					
			1 1,		W 11 (17 (10 D	<u> </u>					
	HOLD	<b>→</b>		<u>}</u>	INVITE(CP						
				<u> </u>	200 OK IN\	/ITE					
			-	<b>&gt;</b>	ACK						
	SETUP	<b>→</b>					<b>→</b>	INVITE(IAM)			
	ALERTING	+					<b>←</b>	180 Ringing(ACM)			
	CONN	<b>←</b>					<b>←</b>	200 OK INVITE(ANM)			
	FAC(est3pty)	→		<b>→</b>	INFO(CPG						
				<u> </u>	200 OK INF	0					
							<b>→</b>	INFO(CPG conf est)			
							<b>←</b>	200 OK INFO			
				3	PTY commu	nication					
	DISC	+					<b>←</b>	BYE(REL)			
	RELEASE	<b>→</b>					<b>→</b>	200 OK BYE			
	REL_COM	+		<b>&gt;</b>	INFO(CGP						
			•	F	200 OK INF						
			-	<b>&gt;</b>	INFO(CGP	(hold)					
			•	<b>F</b>	200 OK INF						
	DISC(CRx)	<b>→</b>	1-	<b>&gt;</b>	BYE(REL)		1				
	RELEASE	+		<b>F</b>	200 OK BY	E					
	REL_COM	<b>→</b>				<del>_</del>					
		tion sh				ne other remo	ote u	ser, followed by the			

TP715006	SIP reference: R			ISUP reference: Q.1912.5 [1], clauses 2.4; 2.2.1/Q.734.2 [i.8]						
TSS reference	ISDN-(ISUP)-SIP/SS/3P	ΓΥ								
SIP selection										
criteria										
ISUP selection										
criteria										
Test purpose	Remote user included in 3PTY  To verify that the IUT can receive the notification information related to 3PTY, and pass it on to the access signalling system. The IUT should be able to transparently transfer the CPG message with the following notifications in the generic notification indicator in both the forward and the backward direction:  1) "Conference established".  2) "Conference disconnected".  3) "Remote hold".									
SIP parameter values ISDN parameter										
values										
Comments	ISDN		SU	IT	SIP					
	SETUP	+		+	INVITE(IAM)					
	ALERTING	→		→	180 Ringing(ACM)					
	CONN	→		<b>→</b>	200 OK INVITE(ANM)					
		С	ommur	ication						
				<b>←</b>	INVITE(CPG hold)					
				<b>→</b>	200 OK INVITE					
				<b>←</b>	ACK					
	NOTIFY(conf est)	+		<b>←</b>	INFO(CPG conf est)					
				→	200 OK INFO					
			Y comn	nunication						
	NOTIFY(conf disc)	+		<b>←</b>	INFO(CPG conf disc)					
				→	200 OK INFO					
	NOTIFY(hold)	+		+	INFO(CPG hold)					
				<b>→</b>	200 OK INFO					
	DISC	+		<b>←</b>	BYE(REL)					
	REL	<b>→</b>		<b>→</b>	200 OK BYE					
	REL_COM	<b>←</b>								

# Annex B (informative): Bibliography

ITU-T Recommendations Q.761: "Signalling System No. 7 - ISDN User Part functional description".

 $ITU-T\ Recommendations\ Q.762:\ "Signalling\ System\ No.\ 7-ISDN\ User\ Part\ general\ functions\ of\ messages\ and\ signals".$ 

ITU-T Recommendations Q.763: "Signalling System No. 7 - ISDN User Part formats and codes".

ITU-T Recommendations Q.1902.1: "Bearer Independent Call Control protocol (Capability Set 2): Functional description".

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

## Annex C (informative): Change history

Date	WG Doc.	CR	Rev	CAT	Title / Comment	Current Version	New Version
10-06-	21PTD096r	001		F	Update of test description and message flows	1.1.1	1.2.1
09	1						
					Publication	1.2.1	1.2.1

### History

Document history		
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